Teemu Koski

Audiometry Using Realistic Sound Scenes Reproduced with Parametric Spatial Audio

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Thesis supervisor:

Docent Ville Pulkki

Thesis instructor:

D.Sc. (Tech.) Ville Sivonen

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School of Electrical
Engineering

AALTO UNIVERSITY
SCHOOL OF ELECTRICAL ENGINEERING

ABSTRACT OF THE
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Author: Teemu Koski

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Supervisor: Docent Ville Pulkki

Instructor: D.Sc. (Tech.) Ville Sivonen

The term audiometry denotes the testing of hearing performance. It is conventionally conducted by measuring the hearing thresholds for pure tones over headphones. However, especially hearing-impaired individuals and hearing instrument users generally have the most problems in communication situations where background noise is present. Results from pure-tone audiometry do not necessarily describe these problems well. To better assess the real-life hearing performance, sound-field audiometry (SFA) systems have been developed, in which loudspeakers are used instead of headphones. Speech and real or synthetic background noise materials are typically used in SFA systems. However, most of the current SFA systems are either too large and complex for clinical environments or do not reproduce the spatial characteristics of the sound scene correctly.

Directional Audio Coding (DirAC) is a parametric spatial sound reproduction technique, which utilizes the knowledge on the temporal and spectral resolution of human hearing. With DirAC, the spatial attributes of sound can be captured and reproduced with arbitrary loudspeaker setups.

In this thesis, DirAC was applied to audiometric purposes. A sound-field audiometry system was proposed, with which speech intelligibility assessments can be done in realistic pre-recorded sound scenes where external test speech is augmented. With acoustic measurements and psychoacoustic listening tests, a comparison was made between a reference sound scene and a reproduced scene where the reference was reproduced by the method under investigation. The main result was that speech intelligibility did not differ notably between the reference and the proposed system. This was confirmed in listening tests conducted with both a group of normal hearing test subjects and a group consisting of cochlear implant and hearing aid users. The results suggested that the proposed method is valid for clinical hearing diagnostics. The main advantage of the system is that it enables the assessments of real-life hearing abilities using a relatively compact loudspeaker setup. Requirements were specified for a clinical implementation of the system, considering the loudspeaker setup and the test room acoustics.

Keywords: Psychoacoustics, audiology, sound-field audiometry, spatial sound, parametric spatial audio, DirAC

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Audiometria tarkoittaa kuulon toiminnan tutkimista. Perinteinen metodi on äänesaudiometria, jossa potilaan kuulokynnys mitataan siniääneksillä kuulokkeita käyttäen. Kuulovammaiset ja kuulolaitteen käyttäjät kuitenkin tyypillisesti kokevat hankalimpina kommunikaatiotilanteet taustamelussa. Äänesaudiometria ei mittaa näitä ongelmia kunnolla. Äänikenttäaudiometriassa käytetään kaiuttimia kuulokkeiden sijaan, jolloin kuulon käytännön suorituskykyä voi mitata paremmin. Tällaisissa järjestelmissä käytetään testimateriaalina tyypillisesti puhetta sekä äänitettyä tai synteettistä taustamelua. Useimmat nykyisistä äänikenttäaudiometriatoteutuksista tosin ovat joko liian suuria ja kompleksisia kliinisiin ympäristöihin tai eivät toista äänen tilaominaisuuksia kunnolla.

Directional Audio Coding (DirAC) on parametrinen tilaäänen analysointi- ja toistotekniikka, joka hyödyntää tietoa ihmisen kuulon aika- ja taajuusresoluutiosta. DirAC:n avulla äänen tilaan liittyvät ominaisuudet voidaan tallentaa ja toistaa mielivaltaisella kaiutinjärjestelmällä.

Tässä työssä sovellettiin DirAC:a audiometriaan. Työssä esiteltiin äänikenttäaudiometriasovellus, jonka avulla kuulon diagnostiikkaa voidaan tehdä realistisissa ennalta äänitetyissä ääniympäristöissä, joihin on augmentoitu ulkoista testipuhetta. Referenssiääniympäristöä ja sen DirAC-toistettua kopiota verrattiin keskenään akustisin mittauksin ja psykoakustisin kuuntelukokein. Päätulos oli, että puheenymmärrettävyys ei poikennut merkittävästi näiden ympäristöjen välillä. Tämä todistettiin kuuntelukokein, joissa koehenkilöinä käytettiin sekä normaalikuuloisia että kuulolaitteen ja sisäkorvaistutteen käyttäjiä. Ehdotettua metodia voi tulosten perustella käyttää kuulon diagnostiikkaan klinikkaympäristössä. Sovelluksen tärkein etu on sen tuoma mahdollisuus mitata kuulon tosielämän suorituskykyä verrattain kompaktilla kaiutinjärjestelmällä. Järjestelmän tekniset vaatimukset määriteltiin kaiutinjärjestelmän ja testihuoneen akustiikan osalta.

Avainsanat: Psykoakustiikka, audiologia, äänikenttäaudiometria, tilaääni, parametrinen tilaäänentoisto, DirAC

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I would also like to give a warm hug to my family and especially my beloved Pauliina for supporting me in whatever I end up doing. Finally, additional hi-fives go to my friends and bandmates for providing me counterbalance during the research process.

Otaniemi, 19.3.2012

Teemu J. Koski

Contents

Abbreviations

BILD	Binaural intelligibility level difference
BMLD	Binaural masking level difference
CI	Cochlear implant
dB	Decibel
dBA	Decibel (A-weighted)
DirAC	Directional Audio Coding
DRR	Direct-to-reverberant ratio
HA	Hearing aid
HINT	Hearing in noise test
HL	Hearing level
HRTF	Head-related transfer function
Hz	Hertz
kHz	Kilohertz
MILD	Monaural intelligibility level difference
MLD	Masking level difference
SFA	Sound-field audiometry
SNR	Signal-to-noise ratio
SPL	Sound pressure level
SRM	Spatial release from masking
SRT	Speech recognition threshold
SRTN	Speech recognition threshold in noise
sSRT	Sentence speech reception threshold
PTA	Pure-tone audiometry
VBAP	Vector Base Amplitude Panning
WFS	Wave Field Synthesis

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Chapter 1

Introduction

1.1 Background

Modern living environments contain various kinds of sound information and noise. This emphasizes the importance of human ear to perform in its fundamental task: enabling communication in various sound scenes. High technology is being applied to hearing instruments to make people cope with hearing disorders. Altogether, this situation emphasizes the importance of reliable and representative hearing diagnostics.

The term audiometry denotes the testing of hearing performance and it can be conducted in several ways. Pure-tone audiometry over headphones is the conventional method: it gives the thresholds of hearing in different frequencies, which is a straightforward measure of how well the auditory system is reacting to sound. However, testing with pure tones does not reflect the real-life hearing abilities: people with hearing impairments and users of hearing instruments generally report to have the most problems in everyday communication, where background noise and complicated room acoustics are present. Furthermore, headphone-listening is problematic and limited in several test cases.

An alternative approach for hearing diagnostics is sound-field audiometry (SFA), where loudspeakers are used instead of headphones. In addition to solving the headphone-related problems, sound-field audiometry allows testing the spatial aspects of hearing, which is essential in assessing the real-life hearing performance. In sound-field audiometry, speech intelligibility in the presence of a masking noise is an important measure. To measure this, setups have been proposed in which speech is reproduced from one loudspeaker and noise from another. However, these kind of two-loudspeaker approaches are limited in giving reliable and representative results. Recently, partly due to development in spatial sound technology, a growing interest has been to simulate real-life sound environments and room acoustics for audiometric applications.

One recently developed technique for spatial audio is Directional Audio Coding (DirAC). It is a parametric spatial sound reproduction technique for arbitrary loudspeaker setups. DirAC is developed in Aalto University among its underlying techniques Spatial Impulse Response Rendering (SIRR) and Vector Base Amplitude Panning (VBAP). Although not applied in the field of hearing diagnostics, these techniques have been shown applicable for example to high-quality multi-channel audio and teleconferencing. Experiments in these

applications have shown good results even with setups consisting of relatively low number of loudspeakers. This motivates the research on the use of these techniques in sound-field audiometry.

1.2 Aim of the thesis

This thesis investigates the use of Directional Audio Coding for hearing diagnostics over a loudspeaker setup simple enough to be used in clinical environments. The initial aim is to design a system for audiometric tests that could be used in clinical environments, bearing in mind the current needs and existing systems. The system would aim for assessing the real-life hearing performance and communication abilities relevant to the patient. Additionally, the functional gain of a hearing instrument could be measured with the system.

Besides the expected advantages, the use of spatial sound techniques in this application highlights several considerations. Namely, DirAC utilizes some knowledge of the resolution of human spatial hearing, and although it is well tested with normally-hearing listeners, the way how of hearing instrument users and hearing-impaired individuals perceive DirAC-reproduction is unclear. Furthermore, the SFA system to be designed should be compact enough for availability to clinical use, but at the same time accurate enough for reliable audiometric testing. That is, a compromise has to be made between system versatility, reproduction accuracy, and the number of loudspeakers used. Indeed, reducing the number of loudspeakers tends to lower the reproduction accuracy. This framework opens up a set of questions, which this thesis aims to answer:

- Is it relevant to conduct sound-field audiometry using real-life sound scenes reproduced with DirAC, while patients include both normally-hearing and hearing-impaired individuals and also hearing instrument users, and how could this be implemented?
- Could the reproduction be done with a setup compact enough to be used in clinical environments, and what would be the technical requirements for the system then?

These questions are discussed through literature study, experimental engineering, acoustic measurements, and psychoacoustic listening tests.

1.3 Outline of the thesis

This thesis divides roughly in two parts: the theory part of chapters 2–4 and the experimental part of chapters 5–7. Chapter 2 contains the principles of sound and hearing as prerequisites for the consequent chapters. Chapter 3 introduces spatial sound technologies and describes Directional Audio Coding (DirAC). Chapter 4 gives an overview of hearing defects, their management, and diagnosis, giving emphasis on sound-field audiometry. In Chapter 5, a concept of a sound-field audiometry system featuring DirAC is developed, described and technically validated. Chapter 6 reports the conducted listening tests further validating the system. Chapter 7 summarizes the results of the experimental work and discusses the advantages, drawbacks and applicability of the proposed sound-field audiometry system. Finally, Chapter 8 concludes the thesis.

Chapter 2

Sound and hearing

This chapter introduces the fundamental concepts of sound and hearing. These are by much preliminary information for the consequent chapters. Sections 2.1 and 2.2 aim at explaining what sound is and how it behaves in different environments. Section 2.3 briefly introduces the human auditory system and the physiological basis for hearing. Sections 2.4 and 2.5 explore the performance, range and limitations of hearing from different aspects.

2.1 Sound as a phenomenon

Sound, as described in [73], has two descriptions. In perceptual context, it means an auditory sensation in the auditory system. Sound can be wanted or unwanted: there is often the wanted signal (e.g., speech) and also some unwanted noise (e.g.,, traffic noise). Physically, sound is longitudinal wave motion in a medium, which causes the auditory sensation. Similarly, noise has two meanings. While in perceptual context noise is any unwanted or possibly harmful sound, physically noise is described as a waveform with random changes in instantaneous amplitude [92]. Signal to noise ratio (SNR) is defined as the level ratio between signal (i.e., meaningful information) and noise (i.e., unwanted sound) and is usually expressed in decibels (dB).

This thesis follows the terminology proposed in [8], as follows. Sound event is the physical sound phenomenon, which can act as a stimulus (i.e., stimulate the auditory system). Auditory event or auditory object is the sound perceived by the listener. Sound scene is the physical environment of sound, whereas auditory scene is its perceptual equivalent. The term sound environment is also used in this thesis to describe sound scenes more generally.

A sound wave is usually generated by some object, such as loudspeaker cone, vibrating mechanically and thus coupling the vibration to a medium, such as air. Other sound generation mechanisms are changing airflow, rapidly changing heat sources and supersonic flow. Due to vibration, air particles move back and forth and thus generate pressure minima and maxima. At fixed distance from the sound source, air pressure is oscillating around the nominal air pressure. The amplitude of this oscillation is the sound pressure at this location. Sound pressure is usually denoted as a value relative to the reference pressure, which is approximately the smallest pressure amplitude that humans can perceive. This

relative measure, denoted in decibels, is called sound pressure level (SPL) and is defined as

$$L_p = 20 \log_{10} \left(\frac{p}{p_0} \right) \tag{2.1}$$

where p is the sound pressure and p_0 is the reference pressure of 20 μPa. The wavelength of a sound wave is the distance over which the wave is periodic, for example the distance between two pressure maxima. Frequency defines how many of these periods fits to one second. The relation between wavelength and frequency is defined as

$$f = \frac{\lambda}{c} \tag{2.2}$$

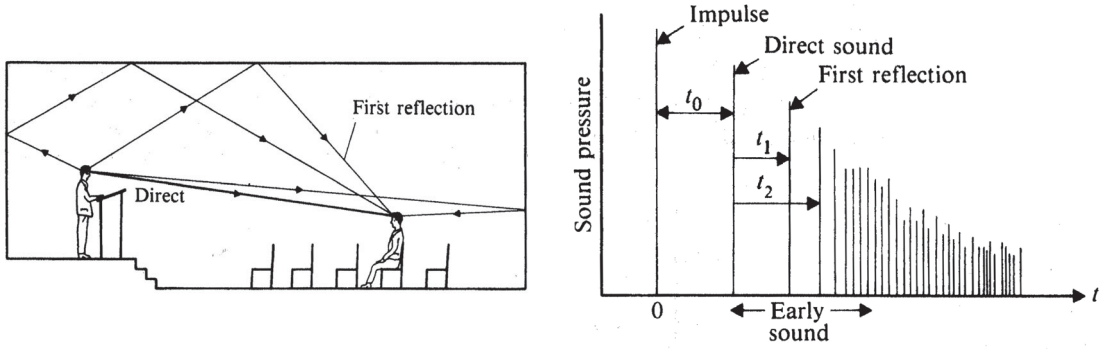
where f is frequency, λ is wavelength and c is the speed of sound. The speed of sound in air is 344 m/s in temperature of 20°C. [40, 73]

2.2 Sound in rooms

As a sound wave generated by a point source propagates freely in space, the sound pressure decreases evenly, with inverse relation to the distance from the sound source. This is because the sound energy supplied by the source is spread to an increasing area as the distance increases. Therefore, sound intensity, which is defined as the sound energy per unit area, is proportional to $1/r^2$, where r is the distance of the sound source. Sound pressure is then proportional to $1/r$. Thus, by Equation 2.1, the sound pressure level of a point source decreases linearly by 6 dB as the distance is doubled. This kind of environment is called a free sound field or free field, where there are no boundaries for the sound to reflect from. The opposite of a free field would be a diffuse field, where the sound is coming evenly from all directions. [73]

In an enclosed space, sound is encountered by objects and surfaces that partly reflect and partly absorb the sound. The amount of how much of the sound energy is absorbed is dependent on the properties of the material and is depicted by the absorption coefficient $\alpha \in [0, 1]$. Due to reflections, sound field in a room – from the viewpoint of the listener – can be separated to three components: direct sound, early reflections and late reflections. Figure 2.1a illustrates the paths of sound in a room. Direct sound is the first wavefront that reaches the ear of the listener. In a free field, this is the only sound that would reach the listener. After the direct sound, the first reflections arrive from the walls, ceiling and floor of the room. The reflections which arrive within the first 50–80 ms after the direct sound are called the early reflections or the early sound. Soon the reflections arrive from all directions and their temporal density is so high that individual reflections cannot be distinguished. These are the late reflections that produce the late sound or reverberant sound. Figure 2.1b illustrates a simplified impulse response of a typical large room or auditorium, where these three components are visible. [73]

Room acoustics affect the perception of sound in a given room. The term room effect is sometimes used to describe the contribution of the room to the resulting sound field and perception. A fundamental parameter in room acoustics is reverberation time (RT). This



(a) Sound reflections in a room

(b) Impulse response of a typical large room

Figure 2.1: Propagation of sound in rooms. Figures adapted from [73].

parameter is defined as the time that it takes for the sound field in a room to decrease by 60 dB [73]. Reverberation time in a room can be calculated with the equation

$$RT = \frac{0.161V}{\sum(\alpha S)} \tag{2.3}$$

where V is the volume of the room and the denominator is the total absorption area of the room, that is, the sum of all surfaces weighted with their absorption coefficients [73]. Another important parameter defining the room effect is direct-to-reverberant ratio (DRR). DRR is the ratio of direct sound energy and reverberant sound energy, typically presented in dB. Additionally, the prominence of the early reflections are often analyzed, while it contributes for instance on the acoustic clarity of the room [5]. Moreover, the direction of arrival of early reflections affects to the spatial impression of the room, or can even distort the localization of sound sources [73].

In addition to reflections and reverberation, sound field in a room is affected by room modes. Room modes are the acoustical room resonances which can be excited by a sound source [73]. These produce peaks and dips to the room magnitude response [73]. Frequencies of the room modes are defined by the equation

$$f_{xyz} = \frac{c}{2}\sqrt{\left(\frac{x}{L}\right)^2 + \left(\frac{y}{W}\right)^2 + \left(\frac{z}{H}\right)^2} \tag{2.4}$$

where x, y and z are integers and L, W and H the dimensions of the room [16].

Calculations with Equation 2.4 show that the mode density increases by frequency. Thus, above a certain frequency, the modes are overlapping and their density is high enough for individual modes to be indistinguishable [77]. This frequency is called the Schroeder frequency, defined as

$$f_c = 2000\sqrt{\frac{RT}{V}} \tag{2.5}$$

where V is the volume of the room [77]. In the context of room acoustics, Schroeder frequency is the crossover point above which a given room is acoustically large and the room modes are negligible [16]. Below that point, the room is acoustically small and the effect of room modes is prominent [16]. According to Equation 2.5, the Schroeder frequency increases as room volume decreases. Thereby, the room modes are essentially an issue of small rooms.

2.3 Auditory system

The human auditory system can be divided in four parts: the outer ear, middle ear, inner ear and central auditory nervous system [92]. The outer ear collects the sound and the middle ear transforms it to vibrations in the fluid-filled cochlea. The inner ear transforms this information to neural impulses, which are analyzed in the central auditory nervous [92]. Figure 2.2 shows a figure of the human ear. Figure 2.3 shows a simplified schematic of the ear.

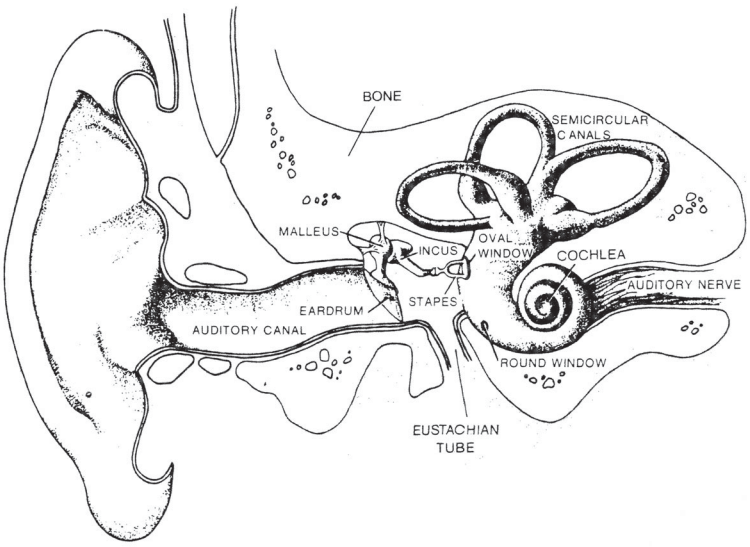


Figure 2.2: A figure of the human ear. The figure is adapted from [90].

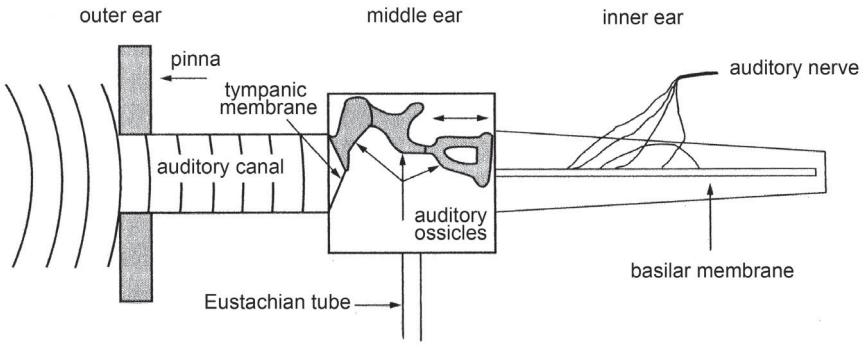


Figure 2.3: A simplified schematic of the human ear. The figure is adapted from [40].

The outer ear is a passive and linear system that collects sound. It consists of the pinna, which is the visible part of the ear, and the auditory canal, which leads to the tympanic membrane, also called the eardrum. The pinna has a complex and asymmetric shape, providing reflections, especially in high frequencies. The auditory canal acts as a resonator with a quarter-wavelength resonance between 2000–5500 Hz. Consequently, the human hearing is relatively sensitive around this frequency range. The outer ear ends to the tympanic membrane, which vibrates with the sound pressure changes in the auditory canal. [40]

The middle ear begins from the tympanic membrane. It consists of three auditory ossicles, namely malleus, incus and stapes, which connect the tympanic membrane to the cochlea of the inner ear. These bones act as an impedance transformer between the air in the outer ear and the fluid in the inner ear, thus enabling the vibration to transmit efficiently. [40]

The stapes in the middle ear is connected to the cochlea in the inner ear through the oval window. The cochlea is a complex auditory organ filled with fluid. It is shell-shaped and has approximately 2.7 turns. The cochlea is often visualized unfolded, as in Figure 2.3, when it would be on average 35 mm long. Figure 2.4 shows a cross-section of the cochlea. Figure 2.5 shows a simplified schematic of unfolded cochlea. The inner ear contains also the vestibular system (i.e., the organs maintaining balance), but it does not affect hearing. [40]

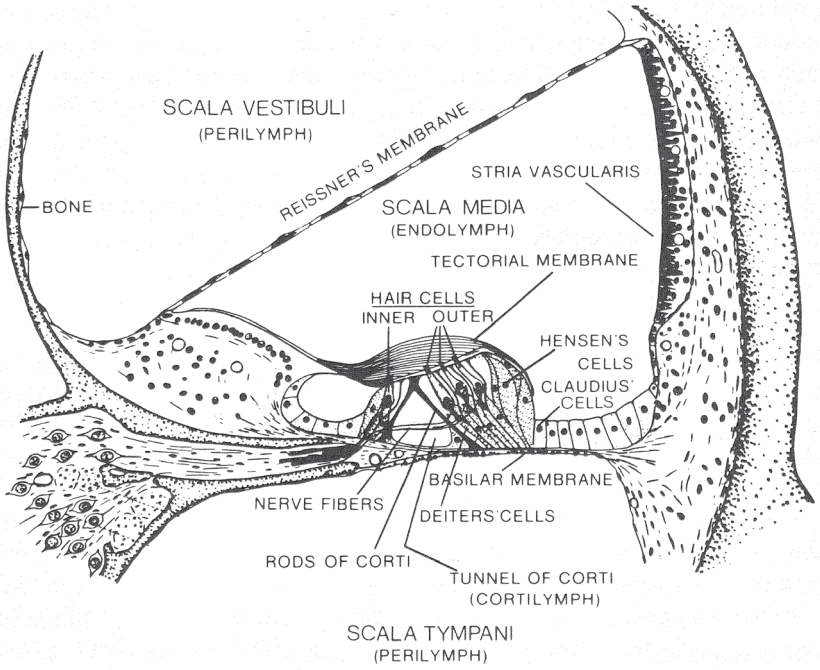


Figure 2.4: A cross-section of the cochlea. The figure is adopted from [90].

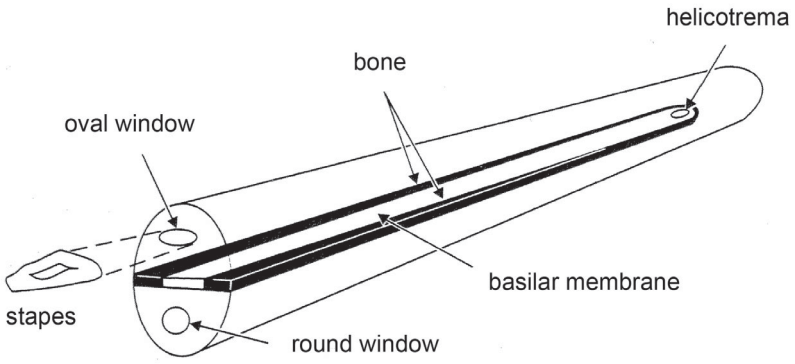


Figure 2.5: A simplified schematic of unfolded cochlea. The figure is adapted from [40].

Vibrations transmitted by the stapes are coupled to the fluid of the cochlea and further to the basilar membrane. In the basilar membrane, there is the organ of Corti, which contains approximately 20000–30000 hair cells, equally placed over the membrane. Two types of hair cells exist, namely outer and inner hair cells. Inner hair cells, total of 3500, are in one row, whereas outer hair cells are in several rows. The hair cells transform the vibration of the basilar membrane to impulses and send them to the auditory nerve fibers. The neural impulse density is proportional to the vibration amplitude of the hair cells, but not linearly: with high vibration inputs, the impulse density output saturates. [40]

The nonlinear transfer function of hair cells shows that signal compression is taken place in the inner ear. According to [42], the outer hair cells act as a biomechanical gain control that provides compression. Furthermore, as stated for example in [42, 56], the outer hair cells react to quiet sound levels and are more fragile than the inner hair cells. According to [41], there are more descending than ascending nerve fibers connected to outer hair cells, whereas to inner hair cells it is the opposite. Thus, the inner hair cells are the primary receptors collecting auditory information, whereas outer hair cells control the movement of the basilar membrane [41].

The basilar membrane can be thought as a spectrum analyzer. The basilar membrane is narrow and light near the oval window, but thickens towards the end [40]. This affects the mechanical impedance seen by a traveling wave in the fluid. Therefore, hair cells are frequency-selective depending on their position on the basilar membrane: the cells in the beginning of the membrane react more to high frequencies and the cells in the end respectively more to low frequencies [40]. This frequency-selectivity is visualized by tuning curves, shown in Figure 2.6. A single tuning curve represents the level of input stimulus needed at different frequencies for a constant output at an individual nerve fiber [40]. Thereby, tuning curves represent the frequency-specific sensitivities of individual hair cells [40]. Figure 2.6 shows that the curves are somewhat wide and overlapping rather than spinous. That is, even a pure tone[1] excites not only one hair cell but an area in the basilar membrane.

[1] A pure tone consists of only one frequency component and it's spectrum is thus a spike in this frequency.

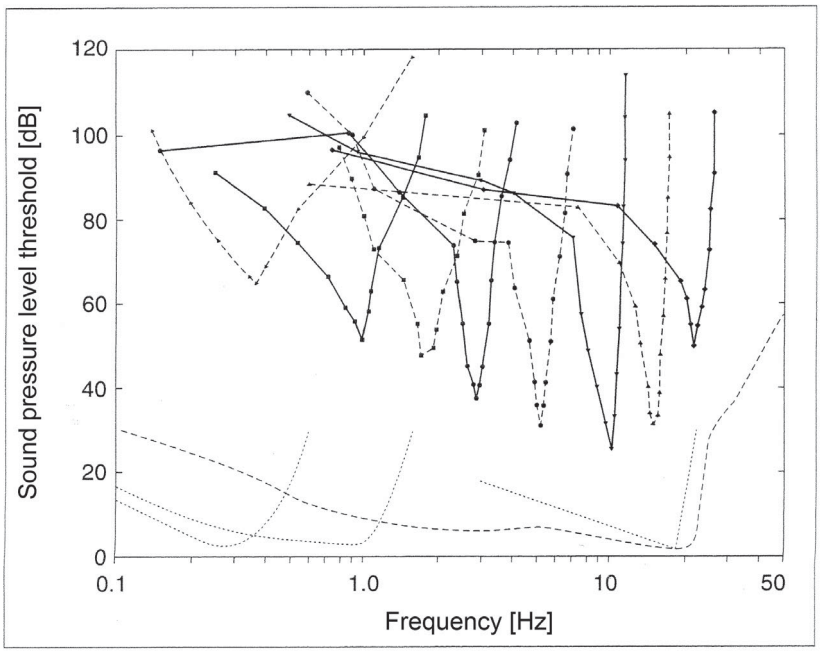


Figure 2.6: Tuning curves measured from individual auditory nerve fibers of cats. The figure is adapted from [40].

From the inner ear, sound information continues to the central auditory nervous system. First, the neural impulses from the cochlea travel to the cochlear nucleus, where it seems that the spectral processing is taken place. The next connection, from the cochlear nucleus to the superior olive, is both contralateral and ipsilateral. Consequently, the superior olive is stimulated by both ears, enabling the analysis of the differences in the signals from both sides. From the superior olive, the pathway continues to the inferior colliculus. However, there are also connections straight from the cochlear nucleus to the inferior colliculus. From inferior colliculus the pathway continues via medial geniculate body to auditory cortex, where the information is interpreted. [92]

2.4 Some attributes of hearing

2.4.1 Sensitivity of hearing

The human hearing range is limited in frequency and level, as visualized in Figure 2.7. In frequency, hearing ranges approximately from 20 Hz to 20 kHz. Sound loud enough below 20 Hz can be sensed as vibration. In level, the dynamic range of hearing is between the hearing threshold and the threshold of pain [73]. Hearing threshold of a young person is approximately $p_0 = 20\ \mu$Pa, which is 0 dB in hearing level (HL) at 1 kHz. Below the threshold of pain, there is loudness discomfort level, above which some distortion of sound may occur [38]. The level of loudness discomfort varies individually.

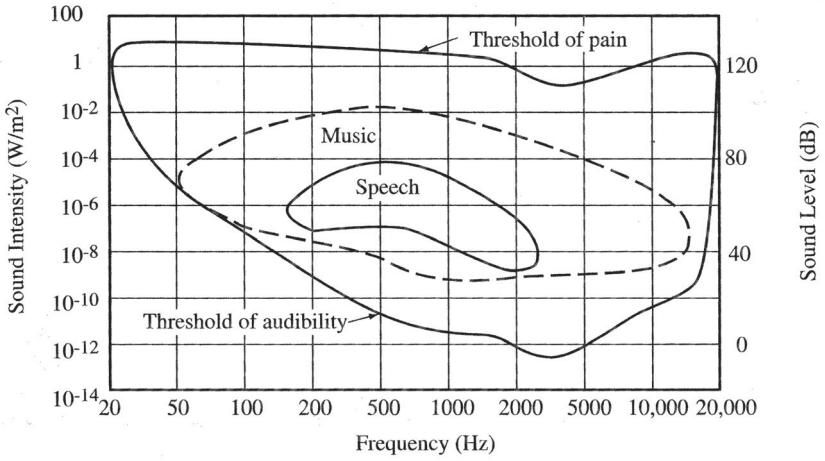


Figure 2.7: The human hearing range in terms of frequency and loudness. The figure is adopted from [73].

2.4.2 Critical bands and masking

The sound scenes of our everyday environment usually consist of not one distinct sound source, but multiple sources competing from attention. Auditory masking is a phenomenon where the perception threshold of one sound event (signal) increases due to the presence of another sound event (masker) [40, 92]. Due to the masking effect, some of the sound events are masked under others so that they are not perceived. Masking occurs both in temporal and frequency domain. Temporal masking is a non-linear, neural-level effect, which is present 5–10 ms before and 150–200 ms after the masker [40]. Frequency-domain masking is of essential importance in the context of this thesis, and is therefore discussed in more detail.

In frequency-domain masking, the effectiveness of a masker depends on its presentation level and frequency content. The masking effect of white noise[2] is visualized in Figure 2.8. When white noise is used as a masker, the masked hearing threshold for a pure tone test signal is constant in frequencies under 500 Hz. Above that, the masking effect increases by 10 dB per decade. The masked hearing threshold – or just masked threshold – basically means the hearing threshold of a test tone when a masking noise is applied.

In the case of a narrow-band noise masker, the masked threshold curves look different. Figure 2.9a presents the masked thresholds for a pure tone test signal, with three narrow-band noise maskers with different center frequencies. The figure shows that masking occurs at a certain frequency band around the masker. Bearing in mind that the frequency scale is logarithmic, one can note that the higher the masker frequency, the broader the frequency range where masking occurs. Another aspect of masking is shown in Figure 2.9b. The figure shows the masked thresholds for a pure tone test signal, with seven narrow-band noise maskers, all with center frequency of 1 kHz but with different presentation levels. As the presentation level increases, the masking effect bandwidth broadens, extending further to higher frequencies. The masking effect produced by a complex tone can be thought as a combination of the masking effects of each of its partial. [40]

[2]White noise is random noise with equal amount of energy in all frequencies.

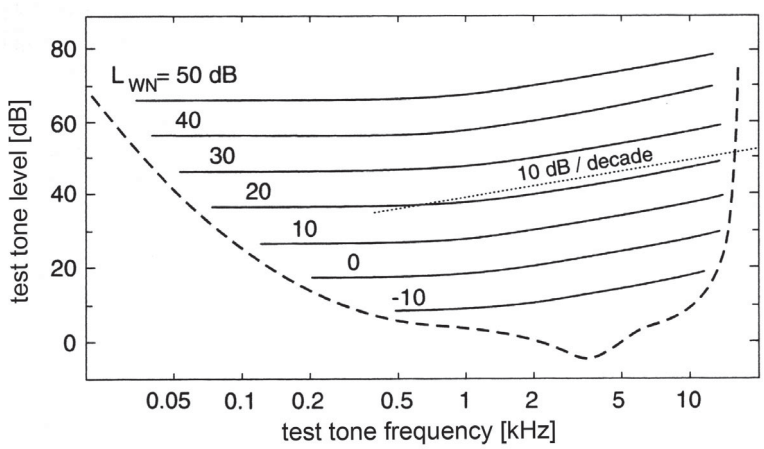
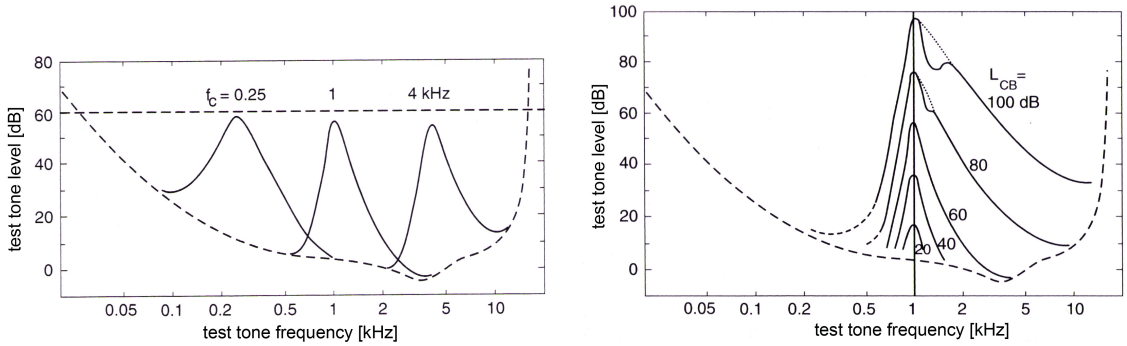


Figure 2.8: The masking effect of a white noise masker for a pure tone test signal. The solid lines represent the masked hearing thresholds when a white noise of different presentation level is applied. The dashed line represents the unmasked hearing threshold. The figure is adapted from [40].



(a) (b)

Figure 2.9: Masking effect of a narrow-band masker with a) different center frequencies and b) different presentation levels. The solid lines represent the masked hearing thresholds for pure tone test signal. The dashed line represents the unmasked hearing threshold. The figure is adapted from [40].

The mechanism of masking can be understood by critical bands. For example, the detection threshold for a sinusoidal signal of frequency f_{test} is dependent on the total energy of a masker on a critical band, with center frequency of f_{test} [92]. The critical bands are not fixed in frequency but formed around any narrow band stimulus [92]. The auditory system processes the contents of each critical band as one entity [40]. This is because the hair cells in the basilar membrane have substantial interaction with the nearby cells [40]. It is thus logical that the tuning curves in Figure 2.6 broaden (in linear scale) as frequency increases, as well as did the masking threshold curves in Figure 2.9a.

The masking discussed so far can be labeled energetic masking. Furthermore, also informational masking occurs. Informational masking is generally non-energetic masking: whereas energetic masking is a process of cochlea and auditory nerve, non-energetic mask-

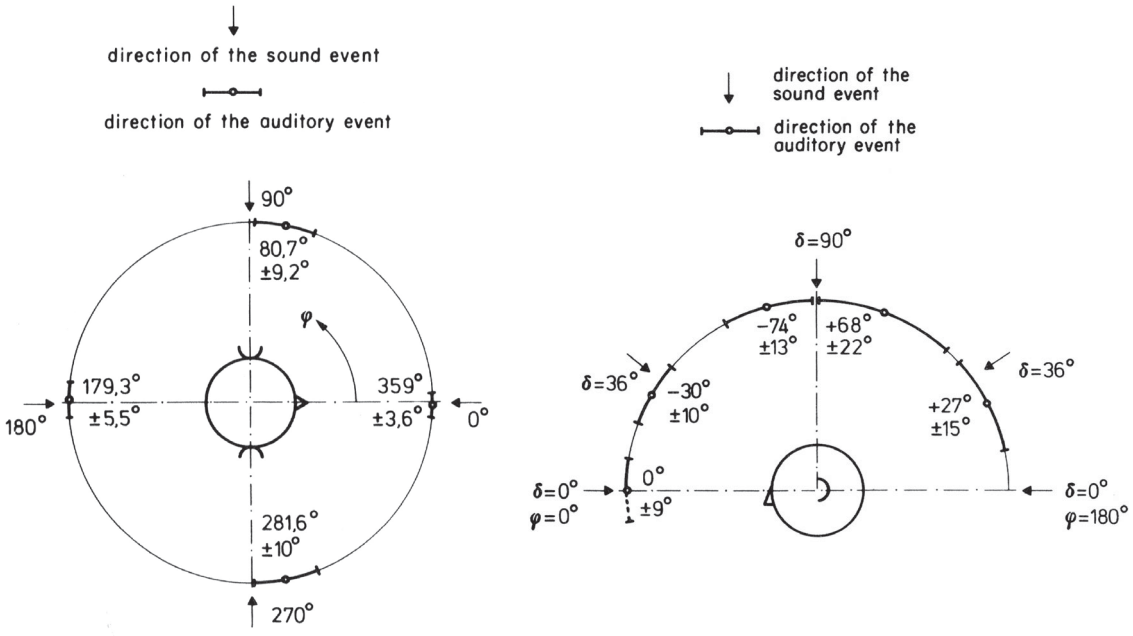
ing is a cognitive process. For example, when speech is masked by speech, the similarity of the signal and masker causes confusion and decreases concentration. [21]

2.5 Spatial hearing

2.5.1 General discussion

The term spatial hearing refers to the aspects and mechanisms of hearing related to direction and surrounding space. Also the term binaural hearing, an equivalent to "hearing with two ears", is often used in this context, while spatial hearing is in large degree dependent on input to both ears.

Directional hearing cabalibities can be quantified for example with the concept of localization blur. As defined in [8], localization blur is "the amount of displacement of the position of the sound source that is recognized by 50 percent of experimental subjects as a change in the position of the auditory event". That is, localization blur describes the accuracy of localization for a sound source at a given direction. Figure 2.10 visualizes the human localization ability in the horizontal plane (i.e., with different azimuth angles) and upper half of the median plane (i.e., with different positive elevation angles), based on studies reviewed in [8]. The figure shows that localization is the most accurate for sound sources in the front. Detecting the elevation of sound sources is much less accurate in general than detecting the azimuth angle.



(a) (b)

Figure 2.10: Human localization ability a) in the horizontal plane and b) in the median plane. The azimuth angle is denoted by φ and the elevation angle by δ. The figure is adapted from [8].

For the following sections, a few terms are elucidated. The term monaural or monotic refers to listening with only one ear, while in binaural condition both ears are stimulated. Binaural listening condition can be diotic or dichotic. In diotic condition, both ears are stimulated with identical signal, while in dichotic condition the signals are different. Unlike in a sound field, in headphone listening sound sources are generally perceived to be inside or nearby the head and therefore it is relevant only to consider the lateral position of the sound sources in the axis of the ears [8]. Thus, as in a sound field localization is discussed, in headphone listening the relevant term to use is lateralization [8].

2.5.2 Binaural localization cues

Two binaural cues contribute to localization of sound sources, namely interaural time difference (ITD) and interaural level difference (ILD) [8]. These cues give information about the left-right direction of the sound source [8]. ITD is created by the difference in the length in the two paths from the sound source to the ears. Due to that difference, sound arrives later to the contralateral ear than to the ipsilateral ear. ILD is created by the so-called head-shadow effect, that is, the head applies an acoustic shadow, that attenuates sound in the contralateral ear. Figure 2.11 illustrates ITD and ILD in practice.

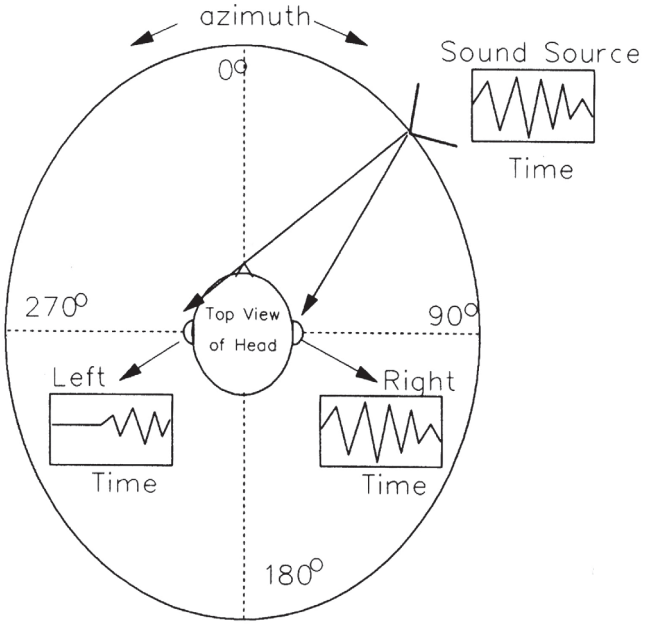


Figure 2.11: Binaural localization cues (ITD and ILD) visualized. Sound source being on the right, the sound signal in the left ear is delayed and attenuated compared to the signal in the right ear. The figure is adapted from [92].

ILD is the main localization cue at high frequencies, where ITD is not perceived [8]. As frequency decreases, the effect of head shadow diminishes, making ILDs small in low frequencies. This is due to diffraction: as the frequency decreases, head dimensions are smaller compared to the wavelength. Thus, perceivable ILDs produced by one sound source do not occur naturally in low frequencies. However, ILDs are perceived in the whole

frequency range of hearing and thus, for example in headphone listening, it is possible to produce perceivable ILDs also in low frequencies.

ITD is the main localization cue in low frequencies. It is also called interaural phase difference (IPD), for in the case of continuous signals time differences translate into phase differences. Experiments with pure tones have revealed that the ability to localize sound sources with ITD decreases rapidly after 800 Hz, and above approximately 1.6 kHz the cue is ineffective. [8]

The diminishing perception of ITD by increasing frequency can be understood by a practical example utilizing Equation 2.2. Considering an approximate distance between the ears being around 17 cm, this is roughly the maximum difference in length in the two paths from the sound source to the ears. Thus, the maximum natural ITD is around 0.5 ms. This distance corresponds roughly to a half wavelength (maximum phase difference) of a one-kilohertz wave. Thus, at frequencies over 1 kHz, IPD can be in between half and one full wavelength. In this case, the cue is confusing: the IPD can be interpreted in two ways depending on which of the signals in ears is considered leading and which lagging in time. Respectively, above 2 kHz, IPD can several cycles, making the analysis even more ambiguous.

Further studies have been made with a high-frequency carrier signal, for example narrow band noise modulated by a low-frequency envelope. Above 1.6 kHz, the auditory system cannot decode the phase differences of the "fine structure" of the carrier, but an ITD applied to the envelope of the signal is perceived. Envelope-ITDs are to some extent detected also under 1.6 kHz, depending on the shape of the envelope. Therefore, the cue produced by carrier and envelope ITDs can be in conflict, resulting in two spatially separated auditory events. [8]

2.5.3 Monaural localization cues

Monaural cues, also referred as spectral cues, are generated by the complex, individual shapes of the pinna, head and upper torso. These form a linear filter, the characteristics of which depend on the direction and distance of the sound source. Mechanisms of this filtering include reflection, diffraction, dispersion, interference, resonance, and shadowing. Thus, spatial information of the sound scene is coded to temporal and spectral cues of the sound signal reaching the tympanic membrane. [8]

Considering the shape of the pinna, the direction-dependent filtering is obvious. Especially the response for sound sources in the back and front differ substantially. Shoulders, in addition to pinna, create a difference to the responses for sources with different elevation. This is essential for the human localization ability, while the use of binaural cues is by much restricted to the lateral position of the sound source. Indeed, ITD and ILD cues are approximately constant in a so-called cone of confusion [91], that can be visualized as a circle formed by the bottom of a cone, the apex of which is pointing to the auditory canal. Without spectral cues, front-back and up-down confusion occurs, while the interaural cues are identical in multiple locations. This means, for example, that a source in frontal hemisphere can be confused to be at the symmetrical location at the rear hemisphere.

2.5.4 Additional factors on localization

Sound source localization is a process with many factors involved. In addition to the cues mentioned in Sections 2.5.2 and 2.5.3, a few other mechanisms are present. First, rotating one's head alters the monaural and binaural cues. This is utilized more or less consciously to fine-tune localization and to solve confusing situations. For example front-back-confusion is solved with head rotation, while the cues produced by sources in the frontal and rear hemisphere change differently when head is moved. Second, what is seen while hearing, contributes on the sound source localization. For example, when a visual cue is present, the auditory event may be settled at that location, although the real sound source is elsewhere. A visual cue may also externalize a sound source in headphone listening. Studies on motional and visual theories are reviewed for example in [8].

Localization provides information on the source direction regardless of the indirect sound present in enclosed spaces. This is due to precedence effect [92]. That is, sound is localized based on the first wavefront (i.e., the direct sound), and the directional information carried by the early reflections is discarded. Precedence effect occurs, when the time difference of the signals is over 1–1.5 ms but under 30–40 ms [40]. For two sound sources with time differences under 1–1.5 ms, the sources merge to one auditory event, which is localized somewhere between the two directions. For time differences over 30–40 ms, the delayed signal distinguishes as an echo [40].

2.5.5 Binaural advantages in communication

Having two ears not only enables sound source localization, but also contributes to the ability to segregate sound sources and to direct attention to one target source. This is crucial in communication situations with background noise, for example in a cocktail party, where one is trying to understand one talker, although many people are speaking at the same time. Indeed, listening in this kind of situations is much easier with two ears. This is called the cocktail party effect [13], the basis of which is on binaural hearing. The advantages of binaural hearing and the phenomena contributing to cocktail party effect are discussed in the following.

Compared to listening with one ear, there are three main advantages in binaural hearing. First, often either of the ears is closer to the talker and has thus a better SNR due to head shadow. Second, binaural cues allow the speech and noise to be processed separately, further unmasking the speech. This is called the squelch effect. Finally, the total loudness is increased, when two ears receive sound. This is called the binaural summation effect. [49]

The cocktail-party effect is partly explained by the concept of binaural masking: the threshold of detecting a signal in the presence of a masker depends on the listening condition. Fundamentally, the thresholds are the same in monotic and diodic conditions, but lower in a dichotic condition. That is, the masking effect is less effective when signal and noise have different interaural configuration. Binaural masking can be quantified with the term masking level difference (MLD). MLD is defined as the difference between the threshold of detecting masked signal in a certain condition compared to monotic condition. MLD can be up to 15 dB, in case of 1000 Hz sinusoid signal masked with white noise. This

suggests, that if one ear is occluded at a cocktail party, the speech intelligibility decreases notably. However, MLD decreases notably when frequency increases. [92]

A term related to MLD is spatial release from masking (SRM). SRM describes the beneficial change in masked hearing threshold when the signal and masker are spatially separated, compared to them being in the same location (i.e., co-located) [50]. Additionally, when speech intelligibility is used as the measured quantity when assessing masking level difference, the terms monaural intelligibility level difference (MILD) and binaural intelligibility level difference (BILD) are used [8].

A study by Marrone et al. [50] exemplifies the concept of binaural masking. In the study, SRM was measured using three loudspeakers, each with their own one-talker speech signal. The target talker was positioned in the front and the two others were symmetrically placed on the sides, with varying azimuth. Due to the signals used, there was both energetic and informational masking involved. The SRM was measured to be 8 dB when the separation between signal and masker was $\pm 15°$. The maximum SRM of 12 dB was reached after $\pm 45°$ and no substantial change was noticed between angles ± 45–$90°$. When one ear was occluded, this effect was gone.

Another conclusion in the study by Marrone et al. [50] was that informational masking had a greater effect in a co-located condition than in a spatially separated condition. This was noticed when SRMs were measured with the masker signals time-reversed, which eliminated the informational masking effect of the speech. Time-reversing increased the performance (i.e., lowered the masked threshold) by 12 dB in the co-located, but only 5 dB in the spatially segregated condition. This means that the SRM is higher when informational masking occurs. As the spatial segregation had already applied a large improvement in the masked threshold, eliminating the informational masking did not further improve the situation as much as it was improved in the co-located condition.

Furthermore, in the SRM-tests of Marrone et al. [50], the SRM decreased when reverberation was added to the test booth. This is logical: as reverberation increases the diffuseness of the sound field, the effective spatial separation of such sources decreases. Similar results were achieved in [72], in which the benefit of hearing aid with directional microphone was studied: it was noted that increased reverberation decreased the directional benefit and performance.

The effect of spatial separation of speech and background noise signal to the speech intelligibility was studied by Rychtáriková et al. [74]. In that study, significant differences were noticed between the speech intelligibility measured in cases S0N0, S0N90, and S0N180 in anechoic conditions[3]. The best intelligibility was generally in the case S0N90, and the second best in the case S0N180. Poorest performance was recorded in the case S0N0, where there was no spatial separation. The results were explained with binaural filtering in the case S0N90 and spectral filtering in the case S0N180. However, when the same test was conducted in a reverberant room, the differences between the three cases were minimal. [74]

[3]The notation SxNy means that the signal and noise are presented in azimuths x and y, respectively.

Chapter 3

Techniques for spatial audio

In the previous chapter, the perceivable spatial attributes of sound were discussed. This chapter briefly discusses techniques for capturing and reproducing such attributes. A cursory overview of spatial audio reproduction techniques is given in Section 3.1. One of such techniques, Directional Audio Coding, is put more emphasis on and described in Section 3.2. Elaborated discussion of these topics are not in the scope of this thesis. Thereby, interested reader is encouraged to explore the references provided in this chapter.

3.1 Approaches to spatial audio reproduction

3.1.1 Spatial audio with loudspeakers

One approach to reproduce spatial audio is Wave Field Synthesis (WFS), in which the physical sound field is targeted to be reconstructed [7]. It is based on Huygens' principle[1]. In WFS, loudspeakers are arranged in an array and each loudspeaker is fed separately with dedicated delay. Loudspeakers must be closely spaced and carefully calibrated. Thus, a very high number of loudspeakers is needed.

Another technique is Ambisonics [26]. In Ambisonics, the sound field in single position is captured with an array of directional microphones and reproduction is done with a loudspeaker array. In first-order Ambisonics, sound is captured in B-format, which consists of four microphone channels. Higher-order approaches utilize more channels. With ambisonics, high-quality reproduction is achieved in the center of the loudspeaker array, in a so-called sweet spot. However, this sweet spot is small: in first-order Ambisonics, the sweet spot is larger than a human head only in frequencies under 700 Hz [81]. Problems arise when listening outside of the sweet spot. Namely, the loudspeaker signals are coherent, resulting in comb-filter effects outside the sweet spot. Furthermore, due to precedence effect, sound event is localized to the nearest loudspeakers.

Currently, in consumer sector, spatial audio reproduction is mainly restricted to standardized loudspeaker setups and their dedicated audio formats. Perhaps the most common of these is the 5.1 system, consisting of five mid/high-frequency loudspeakers and one

[1] Huygens' principle states that any wavefront can be constructed as a superposition of elementary sphere waves [93].

subwoofer. Figure 3.1 shows the ITU-R BS.775-1 [36] specification for the loudspeaker placement in the 5.1 system. For this kind of standardized system, audio content can be created with various recording and mixing techniques. Generally, the idea is to generate an enveloping auditory event with the surround loudspeakers. Several variations of this standard with different number of loudspeakers exists, which usually are based on the 5.1 arrangement. The naming logic is the same: first the number of loudspeakers and then the number of subwoofers separated with a point. Thus, a conventional stereo setup can be called a 2.0 system.

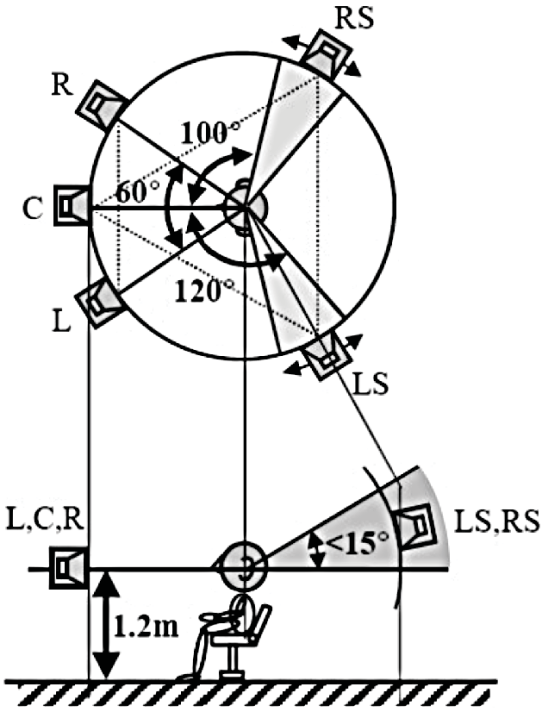


Figure 3.1: The 5.1 standard specified in ITU-R BS.775-1. Letters L, C, R, LS and RS refer to loudspeakers labeled left, center, right, left surround, and right surround, respectively. The subwoofer location is not specified. The figure is adopted from [87].

3.1.2 Spatial audio with headphones

Besides loudspeakers, also headphones can be used in spatial audio. In this case, however, including the monaural localization cues to the headphone signals is required in order to reproduce the sense of space. One approach for this is to do binaural recordings using two microphones placed in the ears of an artificial or real head and reproducing these signals with headphones. Thus, localization cues are naturally coded into the headphone signals. However, when listening to binaural recordings, the sound scene moves as the listener head moves and this decreases the localization accuracy and realism. Furthermore, individual recordings are needed for accurate reproduction.

Another approach is to use Head-Related Transfer Function (HRTF) techniques. HRTFs define how the sound is altered due to the head and shoulders of the listener for given sound

source directions. These functions can be used to produce localization cues in headphone listening [91]. HRTFs are measured using binaural recording methods; generic HRTFs can be measured for example with an artificial head. As well as body shapes, HRTFs are individual. It has been shown for example in [91] that localization performance decreases if non-individual HRTFs are used. Often the problem with non-individual HRTFs is that sound sources are localized inside the head. Using HRTFs is advantageous compared to binaural recordings, because HRTFs can be easily modified and convolved with audio material. For instance, HRTFs can be controlled in real time by listener head movement to enhance the realisticity of the reproduction. Head-tracking is advantageous as it helps to externalize the sound events, that is, to localize the sound sources outside the head instead of inside.

3.2 Directional Audio Coding (DirAC)

3.2.1 The idea in brief

Directional Audio Coding (DirAC) [68] is a spatial sound reproduction method for arbitrary loudspeaker setups. Rather than reconstructing or modeling the physical sound field, DirAC aims at preserving the perceptual attributes of the recorded sound field. Consequently, several assumptions are made of the relation between the physical aspects of sound and the perception they produce. First, the spatial auditory image perceived by a human listener is assumed to be determined by three factors: direction of arrival, diffuseness and spectrum of sound. Thus, if these factors are measured in one location and with the temporal and spectral resolution of human hearing, the spatial auditory image would be reproducible. Second, ITD, ILD, and monaural cues are assumed to define the perceived direction of arrival, whereas interaural coherence is assumed to define the diffuseness of sound. Third, the perceived timbre is assumed to depend on the monaural spectrum as well as ITD, ILD and interaural coherence. Finally, it is assumed that within one critical band and one time instant, the auditory system cannot decode cues from two wavefronts coming from different directions. [68]

Based on the assumptions mentioned above, DirAC analysis and synthesis are implemented as follows. In the analysis, to mimic the spectral resolution of the human auditory system, microphone signals are divided to frequency bands where processing is done separately. Direction of arrival, diffuseness, and timbre of sound are then analyzed with the temporal accuracy relevant to the auditory system in each frequency band. In the synthesis, the signal is dynamically divided to diffuse- and nondiffuse streams which are reproduced in a different manner. The diffuse stream is reproduced by all loudspeakers in use, aiming at surrounding perception of sound with no prominent direction of arrival. The nondiffuse stream is reproduced with Vector Base Amplitude Panning (VBAP) [63] aiming to produce point-like virtual sound sources. [68]

DirAC is based on a technique called Spatial Impulse Response Rendering (SIRR), which is described in [53]. The assumptions reviewed above are common with DirAC and SIRR. In a sense, SIRR is like DirAC for impulses. That is, in SIRR, signal content is known to be an impulse and the recording has been made with one source.

In addition to high quality spatial audio reproduction (e.g., in [85]), DirAC has been

applied to hearing aids [2] and teleconferencing purposes [1]. SIRR on its own enables for example room acoustics reproduction [69].

3.2.2 A-format and B-format input signals

Several implementations of DirAC exist, using either A-format or B-format input signals. A-format is the output format from a microphone array consisting of four cardioid or subcardioid microphone capsules on the faces of a tetrahedron, such as the Soundfield microphone [25]. The A-format capsule signals are denoted LF, LB, RF, RB, which refer to left-front, left-back, right-front, and right-back. An A-format signal can be converted to B-format by linear combination of the A-format microphone capsule signals. B-format consists of four signals: one omnidirectional signal (W) and three figure-of-eight signals pointing forward, left, and up (X, Y, and Z) [26]. B-format signals are used also in first-order Ambisonics [26]. Furthermore, from B-format signal, virtual microphone signals pointing to any direction can be formed by a linear combination.

3.2.3 Limitations and drawbacks

As all spatial audio techniques, DirAC has its limitations and drawbacks. One limitation arises from the non-idealities of VBAP. Virtual sound sources generated with VBAP are sharp and localized accurately when they are positioned near the median plane and generated by a loudspeaker pair or triplet symmetrical to the median plane [66]. However, localization of virtual sources positioned further from the median plane is a bit biased towards the median plane [67]. Also, depending on the source direction, the virtual sources are not always as point-like as real sources [64]. Furthermore, coloration of the virtual sources occurs: for a virtual source formed by two loudspeakers, comb filtering occurs with the first dip in the magnitude response located approximately at 1–2 kHz. [65]. This effect is dependent on the number of loudspeakers used and their positioning [65]. Also, reverberation in the listening room decreases the audibility of the effect [65].

Another issue arises from the psychoacoustic assumptions of DirAC. The validity of these assumptions has not been proved with hearing-impaired listeners or hearing instrument users.

Also, the estimates of DirAC parameters are distorted by the non-idealities of the microphone used. For example, the four microphone capsules of a Soundfield microphone are not coincident, resulting in overestimated diffuseness in high frequencies. However, this problem was addressed for instance in [61], where a DirAC implementation using A-format input signals was described. In the A-format version, the estimation of diffuseness and direction of arrival is correct up to higher frequencies, compared to earlier implementations with B-format input signals [61].

Chapter 4

Overview of technical audiology

This chapter briefly introduces the key concepts of technical audiology in the scope of interest of this thesis, providing the essential prerequisites for the following chapters. Section 4.1 classifies different hearing disorders and discusses how they affect to hearing, the main focus being on sensorineural hearing loss. Section 4.2 discusses hearing instruments. An overall glance to hearing diagnostics is given in Section 4.3. Hearing diagnostics conducted with loudspeakers (i.e., sound-field audiometry) is a key topic in this thesis and is thus discussed separately and in more detail in Section 4.4.

4.1 Hearing disorders

4.1.1 Types of hearing disorders

A hearing disorder is a structural or functional impairment of the auditory system. Most importantly, a hearing disorder degrades communication abilities. Even a slight hearing disorder can affect the perception of music. A more severe hearing disorder complicates the ability to react to sonic signals in the environment. Moreover, sufficient hearing in the early age is of paramount importance to ensure proper linguistic development. [40]

Medically, hearing disorders divide into two main types: conductive and sensorineural. Conductive disorders are due to abnormalities in the outer and middle ear whereas sensorineural disorders are due to abnormalities of the inner ear or the auditory nerve. Sensorineural disorders can be further divided in cochlear (i.e., sensorial) and retrocochlear (i.e., neural) disorders. Finally, although not discussed in this thesis, there are central disorders, which relate to disfunction in the central auditory nervous system. [38, 40, 51]

Hearing disorders result in varying symptoms. The most common symptom is degradation of the sensitivity of hearing (i.e., hearing loss), which can be conductive or sensorineural. In addition to hearing loss, sensorineural disorders include tinnitus and hyperacusis. The following subsections discuss the origin, symptoms and consequences of the hearing disorders mentioned above.

4.1.2 Conductive hearing loss

Conductive hearing losses are caused by abnormalities in the conductive path in auditory system, that is, the outer and middle ear. Outer ear abnormalities can be such as occlusion of the auditory canal due to ear wax, foreign object, tumor, or deformation. Also, the tympanic membrane may become perforated, thickened, or scarred due to mechanical trauma or infection in the middle ear, in some cases leading to hearing loss. [51]

There are several middle ear abnormalities that may cause hearing loss. Infection of the middle-ear cavity is the most common of these, especially in children. Another cause is Mucous otitis media, which is caused by mucoid secretions that have ended up into middle ear via eustachian tube. Also, malfunction of the eustachian tube may cause air pressure difference between middle and outer ear, displacing the tympanic membrane and possibly causing mild hearing loss. [51]

The hearing losses mentioned above usually apply equal amount of attenuation across frequencies. However, some middle ear abnormalities cause hearing loss with non-flat spectrum. Significant negative pressure in the middle ear can cause serous effusion, which means fluid accumulation to the middle ear. This increases the mass of the middle ear, and therefore the resulting hearing loss is more severe in high frequencies. In adults, the most common middle-ear related cause for hearing loss is otosclerosis. It is a progressive disorder where a new growth of spongy bone stiffens the movement of the auditory ossicles, often partially fixing the stapes to the oval window. This attenuates the vibration transmitted by the middle ear. The hearing loss caused by otosclerosis is either flat in frequency or somewhat more severe at low frequencies. [51]

4.1.3 Sensorineural hearing loss

Sensorineural hearing loss is often caused by excessive exposure to noise. Noise-induced hearing loss is caused by a single brief or repeated exposures to high-level sound. The term acoustic trauma is commonly used to describe a brief exposure to high level sound, for example a gunshot. The hearing loss caused by a single acoustic trauma can be followed by complete or partial recovery, although it can result in permanent impairment. Repeating temporary impairments usually result in permanent hearing loss. [40, 51]

In the auditory system, hair cells are the most vulnerable part to acoustic overstimulation. Although a really high level noise can damage also other parts of the ear, such as the tympanic membrane, much lower levels are adequate to damage hair cells. Hair cells can recover from slight damages, but if severely damaged, hair cells do not recover nor are new cells generated. The degree of noise-induced hearing loss depends on the energy received by ear, that is, the combination of sound level and exposure duration. After such damage, the metabolic activity in the exerted hair cell contributes on the permanency of the impairment. [92]

Besides noise, there are several other causes for sensorineural hearing loss. The most usual of them is presbycusis, the aging-related hearing loss. On the other hand, sensorineural hearing loss can be inborn. Furthermore, damages of the inner ear can be caused by a serious head trauma or cancer in the auditory nerve. Some drugs and other substances are ototoxic, meaning that they damage the inner ear. Otosclerosis, discussed in Section

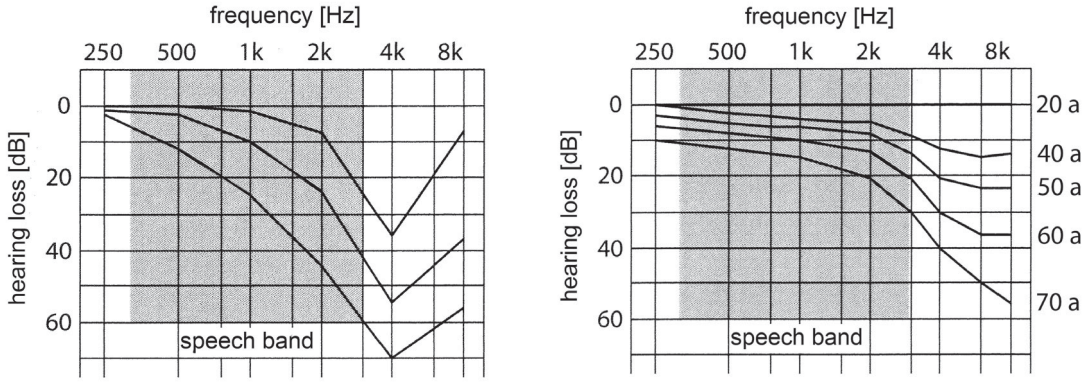
4.1.2, can also involve the cochlea and thereby result in sensorineural hearing loss. Sudden idiopathic hearing loss may occur by viral or vascular origin. Finally, Ménière disease can cause sudden hearing loss. [51]

Difficulty in speech understanding, especially in the presence of background noise, is a common problem among people with sensorineural hearing loss [10, 56, 62]. Indeed, studies have shown degraded speech intelligibility in noise among individuals with sensorineural hearing loss compared to normally-hearing individuals [56]. It appears that people with sensorineural hearing loss are less able to benefit from the temporal and spectral dips in speech [56]. This reflects a disfunction of the active mechanism in the cochlea, that is, the mechanism providing nonlinear characteristics to hearing, such as frequency selectivity and compression [56]. Damage in the outer hair cells inhibits the mechanism to function properly [56]. In the following, the aspects of sensorineural hearing loss contributing to speech intelligibility are discussed.

Hearing threshold shift

Hearing threshold shift is the main consequence of sensorineural hearing loss. It has an obvious effect on speech intelligibility: if a part of the information remains inaudible, speech discrimination is degraded. Depending on the cause, a hearing loss can be equally intense across frequencies or more intense in some frequency range. This contributes to how much the hearing loss affects communication abilities. For instance, discrimination of the fundamental frequency of speech is not as important as hearing the consonants, located in higher frequencies. [38]

Noise-induced hearing loss is often characterized by a notch in hearing threshold at 4 kHz and the poorest hearing in the range of 3–6 kHz [40]. The notch is called the acoustic trauma notch [92]. Typical threshold level curves caused by different amounts of noise exposure are shown in Figure 4.1a.



(a) (b)

Figure 4.1: An example of increased hearing thresholds due to hearing loss caused by (a) noise exposure or (b) aging, in relation to the frequency range of speech. The figure is adapted from [40].

It is important to notice that a mild noise-induced hearing loss does not typically extend to the speech band. However, a severe noise-induced hearing loss affects also the speech band, and thus degrades communication abilities. On the other hand, presbycusis usually begins from the highest frequencies and slowly progresses to lower frequencies with age [92]. Figure 4.1b shows hearing threshold curves typical to patients with presbycusis.

Decreased frequency resolution of hearing

Damaging of the outer hair cells decreases the frequency resolution of hearing [38, 56]. This can be understood with the tuning curves, presented in Figure 2.6. Namely, when the hearing threshold increases, the sharp tip of the tuning curve flattens, making individual curves wider [56]. This makes the hair cell less frequency-selective.

Although the decrease in the frequency resolution itself does degrade the speech discrimination abilities [38], more importantly it affects the masking effect. Namely, as the frequency selectivity decreases, critical bandwidth broadens. Thus, more energy is summed to one critical band and this intensifies the masking effect [38]. In other words, the masking threshold curves shown in Figure 2.9 broaden and the effect of a mask tone spreads to wider frequency area, thus making the masking effect more effective [62].

The effective increase of masking due to sensorineural hearing loss can be up to 10–12 dB [38]. Increased masking effect affects significantly to the speech discrimination abilities in the present of background noise [38]. Therefore, even though speech was above hearing threshold, a person with sensorineural hearing loss generally needs better SNR to recognize speech.

Decreased dynamic range of hearing

Another notable consequence of sensorineural hearing loss is decreased dynamic range of hearing, also called recruitment [83]. In practice this means that the hearing threshold increases but the loudness discomfort level remains. This is because the outer hair cells are damaged but the inner cells are functioning normally [38][1].

The change in dynamic range due to hearing loss is visualized in Figure 4.2. The curves in the figure represent the relation of SPL to perceived loudness, for normal hearing (A), sensorineural impairment (B), and conductive impairment (C). In the sensorineural case, no auditory sensation is present under 40 dB SPL, but on high sound pressure levels the hearing sensitivity is normal. In contrast, the effect of conductive hearing loss is linear: the curve shape is the same as in normal hearing, but biased to the left. This is because both the hearing threshold and loudness discomfort level are increased in conductive hearing loss.

[1] This is consistent with the research reviewed in Section 2.3, where the outer hair cells were stated to act as a signal compressor.

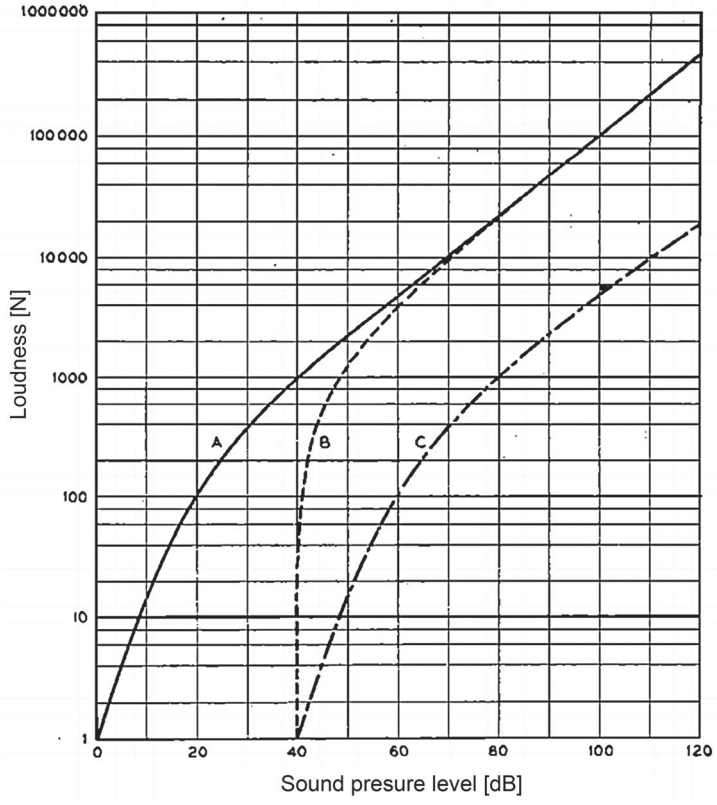


Figure 4.2: Changes in the dynamic range of hearing due to sensorineural hearing loss. The curves show the relation of sound pressure level and the perceived loudness. Curve A is for normal hearing, B for sensorineural hearing loss and C for conductive hearing loss. The figure is adapted from [83].

Additional aspects

Additionally, a reduction of temporal resolution, intensity resolution and temporal integration has also been reported to result from cochlear damage [56]. Moreover, distorted pitch perception and reduced frequency discrimination has been reported [56].

In a case of a severe unilateral or considerably asymmetric hearing loss, the binaural advantages of hearing are obviously degraded or lost. First of all, this causes a decrease in localization abilities [38, 56]. Second, this causes a degradation in hearing abilities in noise due to a decreased masking level difference [56]. That is, the benefit from the spatial separation of signal and noise is decreased [56]. Moreover, for a normal hearing individual, comparing the signals at the two ears, or using the ear with the better SNR, is advantageous when hearing in the presence of noise [56].

4.1.4 Tinnitus and hyperacusis

In addition to hearing loss, several other sensorineural hearing disorders exist. These are such as tinnitus and hyperacusis. Both of these are most commonly caused by excessive noise exposure.

Tinnitus means some kind of auditory event without any external sound event. This "ringing in the ears" can be continuous or occasional. According to [38], over half of the hearing losses are accompanied by tinnitus. However, tinnitus occurs also with individuals with normal hearing thresholds [38]. In some cases, tinnitus is related to a hearing impairment that has not yet realized as a threshold shift [38]. Among the tinnitus patients with acoustic trauma notch, the tinnitus tone can be often matched to the frequency range of 3000–6000 Hz [51]. However, the perceived tinnitus tones vary a lot individually. Besides noise exposure, temporary tinnitus can be due to vertigo, nausea, Méniéré disease and bloodstream-related problems [51].

The literature contains several explanations for the mechanism of noise-induced tinnitus. According to [38], the damaged part of the basilar membrane stimulates the auditory nerve continuously, causing changes to its spontaneous action. The central auditory nervous system does not inhibit this changed spontaneous action [38]. Actually, it may even amplify this, because the sensitivity of the nerve pathways may be dependent on the external sound scene [38]. Hence, the spontaneous action and thereby also the loudness of tinnitus may be increased in quiet environments [38]. On the other hand, several studies have suggested the mechanism of tinnitus to be retrocochlear. For instance, in [6] evidence was shown for tinnitus to be an auditory phantom perception generated in the central level of the auditory nervous system.

Hyperacusis means a decreased loudness discomfort level. While recruitment is fundamentally always preceded by hearing loss, hyperacusis does not necessarily relate to hearing loss. Actually, most individuals with hyperacusis have normal hearing thresholds. However, in hyperacusis the dynamic range of hearing is decreased as the tolerance of loudness is lower. Hyperacusis if often preceded by acoustic trauma and accompanied by tinnitus. [51]

4.2 Hearing instruments

4.2.1 Hearing aid

A hearing aid is like a miniature public address system with signal processing. It amplifies and processes sound to adequately compensate the effect of hearing loss. Different types of hearing aids are shown in Figure 4.3. The small hearing aids inserted into the ear or auditory canal (Figures 4.3a and 4.3b) are advantageous, because they allow the pinna-related directional cues and are cosmetically indistinguishable. The increase in available processing power has enabled the use of more complicated algorithms and more compact size of the device.

The simplest possible hearing aid would be a combination of a microphone near the ear, a linear amplifier with constant gain thorough the audible frequency range, and a miniature loudspeaker providing the amplified sound to the auditory canal. This may be enough to compensate a mild conductive hearing loss with flat spectrum, but not adequate with cases involving frequency- and intensity-dependent attenuation. Thus, modern hearing aids include frequency-dependent amplification, compression and other signal processing, tuned individually for each user. These features are specified in the following.

(a) Behind-the-ear aid (b) In-the-ear aid (c) Completely-in-the-canal aid

Figure 4.3: Different types of hearing aids. The figure is adopted from [60].

Automatic gain control

Automatic gain control (AGC) is beneficial when the hearing loss is level-dependent. AGC was proposed already in 1937 in [83]. AGC is basically a compressor: it enables low intensity sound to be amplified more than high intensity sound, so that the dynamic range of the sound event is compressed. This is helpful in the case of decreased dynamic range of hearing, when the so-called natural compressor of outer hair cells is not functioning properly.

Feedback cancellation

If the signal from the loudspeaker of a hearing aid reaches its microphone, acoustic feedback occurs. Feedback occurs the most likely with small hearing aids, in which the microphone is very close to the loudspeaker [51]. Feedback may occur also due to loose mechanical fitting of the hearing aid or high gain levels [86]. Feedback cancellation algorithms aim at suppressing this feedback. Modern methods are adaptive, that is, they continuously keep track of the loudspeaker output [86]. An adaptive method presented in [82] applies also linear prediction of the upcoming signal to better keep the hearing aid output in the desired range.

Noise suppression

Plain amplification amplifies both signal and noise. Therefore, it does not solve the problem of degraded speech intelligibility in the presence of background noise often faced by individuals with sensorineural hearing impairment. Thus, several algorithms are aimed to increase the SNR in hearing aid output. These help to retain intelligibility in acoustically complex environments, such as noisy or reverberant ones.

Speech recognition in noise can be eased with a directional microphone: when the beam of a directional microphone is directed towards the desired sound source, other directions are relatively attenuated. In beamforming techniques, the signals from multiple microphones are combined to boost or attenuate incoming sound signal form some directions [86]. Some beamforming algorithms are adaptive, so that they can, in some extent, keep track of the directions of wanted and unwanted sound [86]. Blind source separation aims to automatic segregation of different sound sources from the signal captured by the hearing

aid microphone [86]. Beamforming and blind source separation are helpful in the presence of directional noise sources for they can be effectively attenuated. The increased speech intelligibility due to use of a directional hearing aid microphone was studied for example in [15]. The study discovered, for example, that with omnidirectional microphone, a 5 dB better SNR was needed to achieve the same level of speech intelligibility compared to supercardioid [15].

Noise suppression can also be done with a single-microphone devices. Single-channel noise reduction systems use the temporal, spectral and statistical information of the incoming sound and are thus the most effective against stationary and diffuse noise. [62]

Binaural processing

Even if hearing aids were used in both ears, the signal processing in the two aids are not generally interrelated [88]. Consequently, ILD-cues are distorted by the unsynchronized gain processing in the two aids [4]. Furthermore, ITD-cues are often distorted by the monaural noise suppressing algorithms [44].

The following terminology is often used to discriminate between linked and non-linked processing. Bilateral hearing aid refers to using of hearing aids in both ears. Binaural hearing aid instead includes that the processing in the two aids is linked to some extent.

With binaurally aided hearing, binaural cues could be preserved to some extent [62]. This would enhance hearing-in-noise performance, due to detection of spatial separation of sound sources. Preserving of ILD cues in aided hearing was discussed in [4]. In that study, SRT-scores were measured with bilaterally synchronized and unsynchronized gain processing using spatially segregated signal and noise sources [4]. Somewhat better SRT-scores were measured when using bilaterally synchronized hearing aids [4]. Another advantage in binaural processing is that the coherence of left and right input channels can be analyzed and used to suppress diffuse background noise [86]. Additionally, blind source separation could be enhanced with binaural processing [62].

4.2.2 Cochlear implant

A cochlear implant is a bionic ear: it is an electronic device that stimulates the auditory nerve with an electrode array inserted in the cochlea. Thus, the whole conductive and sensory path of the auditory system is bypassed. Alike are bypassed the functions provided by the inner ear, such as frequency analysis and compression.

Figure 4.4 shows a schematic of a cochlear implant. First, there are the external parts, usually located behind the pinna: a microphone, a sound processor, and a transmitter coil. The external parts capture sound, encode it to digital format, and transmit it to the implant. Under the skin, there is a receiver coil and a processor converting the digital sound to electrical impulses. These pulses are sent to an electrode array inside cochlea. Each electrode represents a given frequency band, and stimulates the respective nerve fibers. [14]

The first cochlear implants included only one electrode and were aimed to provide sound awareness. Modern implants have currently up to 22 electrodes, providing more spectral

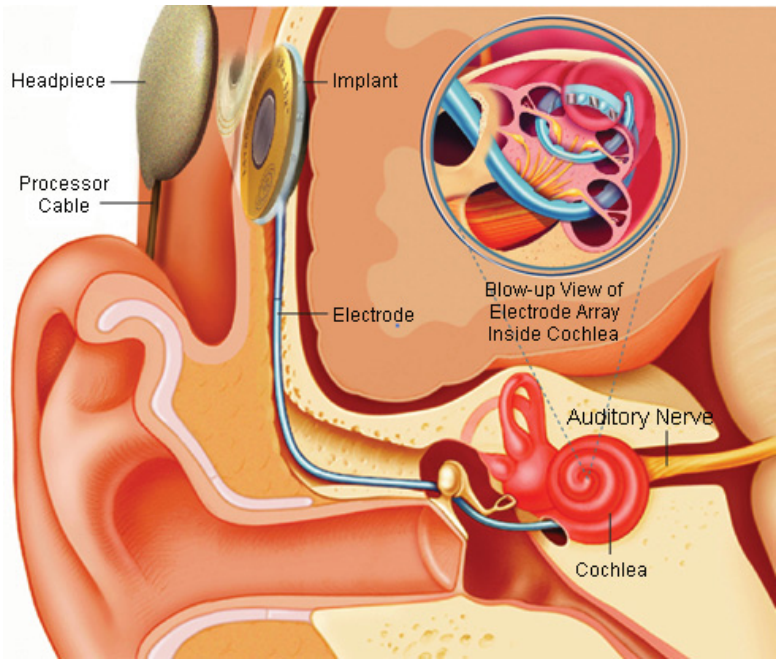


Figure 4.4: A schematic of a cochlear implant. The figure is adapted from [84].

information [14]. The results vary depending on the individual: not every patient reaches good speech recognition, but for some, normal telephone conversation is possible [55].

Although the sound quality in current cochlear implants is not comparable to normal hearing, they can enable adequate hearing in situations where unaided hearing abilities are very low and hearing aid would be useless. This would be the case for example in a severe impairment of the conductive path or severe cochlear impairment.

4.2.3 On hearing with hearing instruments

In aided hearing, some of the problems induced by hearing loss may remain and some new ones may arise. To begin with, hearing through a hearing instrument has an effect on the monaural cues. If the microphone is located behind the ear – or generally anywhere else than in the auditory canal – the spectral cues provided by the pinna are lost. Also, according to [56], the monaural cues are often lost when using hearing aid, because the aid modifies the spectral content of the sound and does not always amplify frequencies above 6 kHz, where the pinna cues are the most prominent. However, some signal processing technologies exists that aim to preserve spectral cues [80].

The effect on the binaural cues is more complex. As discussed in Section 2.5, a normally-hearing person relies on ITD in low frequencies frequencies and ILD on higher frequencies. The research reported in [23] explored the localization mechanisms used by bilateral cochlear implant users. The study participants seemed to rely mainly on ILD. Little supplement was got from ITD cues of the signal envelopes if the test tone was pulsating. Thereby, the localization abilities of the test subjects were quite poor in low frequencies, where ILD cues are not present. In addition, while normally-hearing individuals have two

cues in use for confusing situations, cochlear implant users must get along mostly with one cue. In the study, this was proposed to be a significant issue especially in situations with several sound sources or speech in competing noise.

The relative contribution of ILD and ITD cues to bilateral cochlear implant users was further discussed in [3]. The study concluded that ILD is the major cue used by bilateral cochlear implant users. However, the performance with only ITD cues available was better than with no cues at all, suggesting that some information about the time differences is anyhow used. The performance with ITD and ILD cues was similar to the performance with only ILD cues.

Using two hearing instruments instead of one seems to result in better hearing performance. In [75] it was concluded that binaural cochlear implant users can benefit from the same advantages of binaural hearing to speech intelligibility than do normally-hearing individuals. In [57] the benefit of bilateral cochlear implant to localization and speech discrimination in noise was investigated. In that study, 67 % of the patients showed significant increase in these abilities when a bilateral cochlear implant was inserted, compared to the use of an unilateral implant [57]. Also in [59] the sound localization among cochlear implant users was investigated. That study similarly found significant improvement in the localization accuracy when bilateral implant was used, compared to unilateral implant [59]. In that study, the mean deviation between real azimuth and response was 16.6° with bilateral implant and 53.1° with unilateral implant [59].

However, even if two hearing instruments were used, ITD cues might be lost due to unsynchronized processing in the right and left side. If the processing in the left and right implant was synchronized, the ITD cues could possibly be preserved and thus the hearing performance would possibly be better.

4.3 Hearing diagnostics

4.3.1 Pure-tone audiometry

In pure-tone audiometry (PTA), hearing thresholds are determined at different frequencies using pure tones as test signals. The test procedures in PTA follows an adaptive procedure, defined in standard ISO 8253-1 [32]. Generally, the patient listens in a quiet environment and gives a signal (e.g., presses a button) each time a sound is heard. The test conductor plays short-duration test tones one at a time, alters the sound level depending on the patient's response, and searches the hearing threshold level. This procedure is repeated for each tested frequency. If the patient has tinnitus or a hearing instrument that suppresses pure tones, warble tones can be used as the test signal instead of pure tones [51]. Warble tone is a frequency-modulated pure tone, the frequency of which wobbles rapidly around the tested frequency.

The reference level of normal hearing has been achieved with large-scale measurements of young individuals and is defined in the standard ISO 389 [31]. This curve is called hearing level (HL) and a deviation from it due to hearing loss can be denoted in decibels relative to hearing level (dB HL).

PTA can be done in air or bone conduction. Air conduction measurements are typically

done with headphones and they indicate the degree of hearing loss, but does not specify whether the impairment is conductive, sensorineural, or both. In bone conduction measurement the test tones are reproduced with a vibrator placed on the head behind the ear and the auditory system is stimulated via skull vibrations. Hence, the auditory canal is bypassed and the effect of conductive hearing impairments are disregarded. Thereby, bone conduction measurement determines the sensorineural hearing sensitivity. Air-bone gap (ABG) indicates the amount of conductive involvement in hearing defect. ABG is calculated as the difference between air-conduction threshold and bone-conduction threshold. For example, in the case of zero ABG, the hearing loss is sensorineural. [51]

The results from PTA are visualized in an audiogram, shown in Figure 4.5. The test frequencies usually include the range 125 to 8000 Hz for air conduction mode and range 250 to 4000 Hz in bone conduction [51]. Sometimes the results are wanted to be presented as a single-number average, indicating the overall hearing ability [51]. Several approaches to this exist, one of them being the better ear hearing level, which is the average of the thresholds in frequencies 500, 1000, 2000 Hz ($BEHL_{0.5-2kHz}$), or also with 4000 Hz included ($BEHL_{0.5-4kHz}$) for the better-hearing ear [22]. It is important to notice that the average values can be misleading.

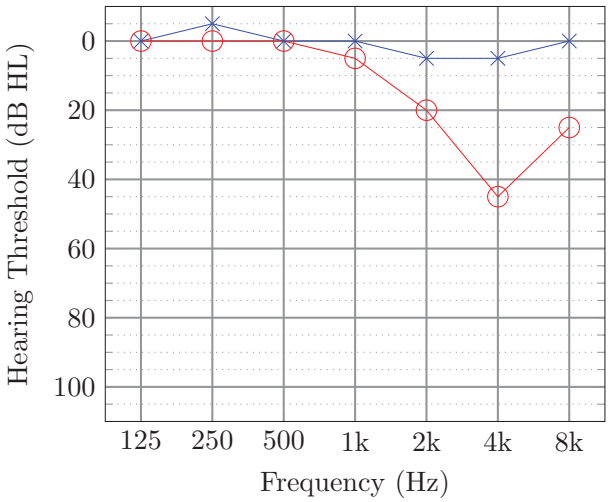


Figure 4.5: An audiogram visualizing hearing thresholds obtained in pure-tone audiometry. The red curve with circles represents the right ear and the blue curve with crosses represent the left ear. This audiogram indicates normal hearing in the right ear and moderate, very likely noise-induced, hearing loss in the left ear.

When conducting PTA, it is essential that the background noise level in the test booth is below the level of masking that could distort the measured thresholds. Closed-back attenuating headphones are thus used to provide some attenuation. Cross-hearing, that is, hearing the test tone with the non-test ear, might occur in case of substantial interaural difference in the hearing thresholds. This can be prevented by applying masking noise to the non-test ear. [51]

Békésy audiometry is an alternative way to implement PTA. In this method, a continuous

pure tone is used and the audible frequency range is sweeped slowly. During the sweep, the patient controls the intensity of the tone with a button: when the button is pressed down, the level decreases and when not pressed, the level increases. With this procedure, a continuous hearing threshold over the hearing range is obtained. [51]

4.3.2 Speech audiometry

Problems faced by the hearing impaired often relate to speech understanding. Rather than giving a frequency-specific measure, speech audiometry aims at assessing the speech intelligibility. Several methods for conducting speech audiometry exists, differing in the measured quantity and the speech material used. The standardized procedures of speech audiometry are defined in the standard ISO8253-3 [34].

Speech intelligibility is commonly measured as the speech-recognition threshold (SRT), also called the speech-reception threshold [51]. SRT denotes the required sound pressure level (usually A-weighted) for the patient to detect 50 % of the speech material presented [58]. Thus, the lower the SRT, the better the speech intelligibility. SRT is typically measured with an adaptive testing method, where the presentation level of the test speech is altered depending on the responses of the patient, and finally the presentation level converges to the SRT [51]. Step size for the presentation level is usually 5 dB, which was proven in [12] to be as accurate as 2 dB in clinical applications. Adaptive testing methods in general are more discussed for example in [48].

In addition to SRT, also other measures for speech audiometry exists. Speech-detection threshold (SDT) is the lowest sound pressure level for the speech stimuli at which the listener detects it as speech. SDT is measured using so-called cold running speech, which means a continuous and monotonous speech material. SDT is generally lower than SRT, when only detecting but not understanding is required. Even more tests exist, such as uncomfortable level test, most comfortable level test, speech-recognition score and word-recognition score. [51]

There are several types of speech material to be used as test tones in speech audiometry, including single words and sentences. The words can be for example spondaic words (i.e., words with two equally-stressed syllables), consonant-nucleus-consonant words, or high-frequency-emphasis words. The sentence material can consist of informative or nonsense sentences. In the latter case, contextual cues are minimized. The term sentence speech recognition threshold (sSRT) can be used when SRT is measured using sentence material. [51]

A dedicated speech material set, a speech corpus, is usually divided in lists, each with a collection of words or sentences that can be used in the same test run. The lists in the corpus are of identical difficulty and loudness so that, for example, the test can be repeated with another list from the corpus. Also, the lists of are often phonetically balanced to provide the proportion of phones that is typical for a given language [51]. Thus, the list imitates normal conversation [51]. It was pointed out for example in [51, 58] that the speech material used as test tones should be of equal difficulty throughout the test. It should be also ensured that learning of the words does not alter the difficulty during the test [58]. This can be done either by familiarizing the listener to the material or using a large enough speech corpus [58].

The literature contains various views on which kind of test material to use. According to [58], individual words do not necessarily represent natural communication due to the lack of natural level fluctuations, intonations, and pacing. Furthermore, the duration of individual words may be shorter than the adaption time of hearing instrument signal processing algorithms [58]. In addition to [58], testing speech recognition with sentence material was proposed also in [39]. However, the use of sentence material in speech audiometry has also been criticized, because if the sentences have meaning, the use of contextual cues affects the intelligibility [51]. Moreover, there seems to be substantial differences in how much different individuals utilize contextual cues [51].

An alternative method to speech audiometry is Coordinate Response Measure (CRM), introduced in [9]. The speech material in CRM consists of a closed set of English words, from which sentences are formed with a certain pattern, for example: "Ready baron, go to blue five now". The patient connects the right number to the right color in a visual display corresponding to what was heard. A clear benefit in CRM is that memorizing the words has no effect on the test. Additionally, the lack of contextual cues and memory effects increase the reliability of the test. [9]

Regardless of the method used, speech audiometry is generally a binaural test. That is, the measured quantity, such as SRT, is achieved using both ears. Thus, the measured speech intelligibility represents the overall hearing performance but gives no ear-specific information.

The relation of pure-tone audiometry and speech audiometry was discussed in [11]. In the study, statistical comparison was made between the results from binaural speech intelligibility measures and results from pure-tone audiometry measured from the better-hearing ear of the same test subject. The results were consistent, although not identical. The conclusion in [11] was that speech intelligibility is a relevant measure, at its best accompanied with pure-tone audiometry, as the combination gives a good overall estimate and shows the consistency of the patient.

4.3.3 Testing speech intelligibility in noise

While the sound pressure level of normal speech at one meter distance is around 60 dBA SPL [73], listening to quiet speech in silence – which is the case in conventional speech audiometry – is not the most natural situation. Instead, the challenging real-life situations often involve background noise. A common complaint by hearing instrument users is that their perceived hearing ability in real-life situations is worse than what is diagnosed in audiometric tests conducted in silence [87]. Thereby, several tests are developed for assessing speech intelligibility in competing noise.

A common approach is to measure SRT in the presence of background noise. Also the term speech recognition in noise (SRTN) can be used is this case. Here, SRT is defined in dB SNR (instead of dB SPL as in tests conducted in silence), where SNR denotes the ratio of the speech signal SPL to the masker SPL [58]. In addition to SRT, many of the other speech audiometry measurements described in Section 4.3.2 can be modified to be used with competing noise. One of the alternative methods is percent intelligibility test, in which the rate of correct detection is measured with constant SNR [58].

The variety of available test speech material was already discussed in the previous section. Another debate is about which masker to use. Indeed, several approaches for the competing noise signal has been proposed in the literature.

First, there are several versions of synthetic noise, such as white noise, equal-loudness noise, speech-shaped noise, and speech-modulated noise. White noise has equal energy at all frequencies while equal-loudness noise is filtered to follow the human hearing sensitivity to provide equal loudness in all frequencies. Speech-shaped noise and speech-modulated noises are filtered to match the long-term average spectrum of speech. In addition, the amplitude of speech-modulated noise varies in time, imitating the temporal level fluctuations of real speech. For example Nilsson et al. [58] suggested the use of speech-shaped noise, because it ensures a constant SNR in all frequencies. Preferably, the spectrum matching should be done with the test speech used in the test [58]. It was pointed out for example in [71] that stationary noise maskers do not represent real-life maskers, because they do not involve the fluctuating spectrum and level.

Second, real speech can be used as a masker. When using speech as a masker, both energetic and informational masking may occur [28]. Often a speech babble is composed, by overlapping many talkers. Time-inverse speech babble can be used, if the effect informational masking is wanted to be disabled [50]. Alternatively, if enough overlapping talkers are used in the babble, informational masking is avoided and the high temporal variation of the amplitude envelope is reduced. However, an argument was made in [70] that speech babble consisting of individual talkers not talking to themselves is unnatural in an acoustic sense compared to real conversation.

The third option is to use environmental noise as the masker. It is a more natural masker than for example shaped noise: it is something that the patients are already used to listen. It seems logical that when real-life hearing performance is assessed, also real-life masker is used. However, bearing in mind the psychophysics of masking discussed in Section 2.4.2, using environmental noise masker is somewhat complicated. The spectral and temporal envelope of environmental noise varies with respect to time, leading to time-varying masking and thereby to time-varying SNR in the test. Hence, the repeatability and reliability of the test may be poor. Also, the spectral and temporal envelope is also different in different noise types. Thereby, an inside-a-car ambience and an in-a-cafeteria ambience produce different kind of masking. This makes it irrelevant to compare speech intelligibility results between different environments without some kind of correction. On the other hand, in [15] the long-term average spectrum of cafeteria noise was reported to be equivalent to speech-shaped noise.

All in all, when selecting the masker to use in speech audiometry, a compromise must be made between the realism of the masker and the reliability of the test. A real recording from a cafeteria consists not only of overlapping talkers but also of several non-stationary sound events. For example, when a loud knock happens in a middle of a word to be listened, speech intelligibility for that word is significantly lower than on average.

One widely used procedure utilizing sSRT in speech-shaped noise is called Hearing In Noise Testing (HINT), which was introduced in [58]. The test uses its own corpus of test sentences: 25 phonemically balanced lists, each with ten American English sentences, balanced in terms of naturalness, difficulty, and reliability. The sentences are based on earlier developed Kamford-Kowal-Bench (BKB) sentences [58]. The forming procedures

and validation of the HINT-sentences are described in [58]. HINT-sentences are intended to be used either without masker or with speech-shaped noise. In [58], the average SRT with HINT was measured to be 23.91 dB SPL without masker and -2.92 dB SNR with competing noise, which was filtered to match the long term spectrum of the HINT-sentences. For the latter case, 72 dB SPL noise was used, meaning that the SRTN was 69.08 dB SPL.

As the HINT procedure and test material are strictly defined, tests done in different clinics should be comparable. The initial idea of HINT has been further developed by many. For example, a set of HINT-sentences in Swedish was developed in [27] with methods similar to the original. The long-term average spectrums of different languages were compared in [18], and accordingly, the differences between English and Swedish are minor. Thus, it was argued in [27] that due to the results found in [18], a Swedish version of HINT sentences could be made that would give results comparable to the international studies using English HINT material.

Currently, there is no official HINT-sentence material produced in Finnish. However, for 50 untrained Finnish university students, a long-term average spectrum of voice was analyzed in [47]. A method comparable to HINT with Finnish speech material was developed by Laitakari [45]. The SRTN test by Laitakari [45] was tested with large number of participants in [46].

In addition to HINT and its modifications, also other procedures exist. The speech perception in noise test consists of eight lists of 50 sentences where the last word of each sentence is considered the test item. The test items are either predictable or unpredictable from the sentence context. For the masker, speech babble with varying SNR is used. In Words-in-Noise (WIN) test, monosyllabic words without linguistic content are used. There, SRT procedure with speech babble masker is used. The WIN test is especially used when the use of contextual cues are wanted to be eliminated. The speech in noise test contains five sentences with five key words for each test condition. In that test, discrete signal levels of 40 and 70 dB SPL are used, performance-intensity functions are calculated, and speech babble with four different SNRs is used as the masker. [43]

4.4 Sound-field audiometry

4.4.1 Advantages of sound-field audiometry

Sound-field audiometry (SFA) is audiometry conducted with loudspeakers. Compared to headphone-conducted audiometry, SFA demands more from the equipment and facilities, but in turn, it enables many test conditions that cannot be implemented with headphones. The motivation for SFA is by much due to the various limitations and problems in conventional headphone-conducted audiometry.

Listening with headphones is impractical for small children and hearing instrument users. Depending on the hearing instrument type and microphone placement, it is often difficult to achieve a constant and controlled acoustic coupling between the headphones and the microphone. Headphones are also problematic when assessing the hearing abilities of children, while they might not tolerate the use of headphones, again resulting in uncontrolled acoustic coupling. Although the acoustic coupling could be achieved, PTA over

headphones does not represent real-life hearing situations. In contrast, testing in a sound field takes into account the spatial attributes of sound and hearing. Additionally, pure tones are attenuated by the feedback control and noise suppression algorithms of hearing instruments and the results may thus become distorted. Finally, testing of hearing in noise and localization is limited with headphones. SFA in turn is more versatile in this sense, because test signals and maskers can be positioned more freely and all localization cues are preserved. For example, testing in sound field is required when comparison is made between listening with own ears versus listening with a hearing instrument [79].

Generally, clinicians have reported that the conventional methods of assessing hearing abilities does not always reveal the underlying problems of the hearing impaired [70]. This is especially true for the problems faced in complicated listening environments [70]. With SFA, it is possible to simulate these environments and thereby assess the hearing abilities in real-life situations.

4.4.2 Test methods in sound field

There are several audiometric tests that can be conducted in a sound field. Some of the most relevant of them are discussed here.

Speech audiometry in sound field enables various kinds of speech intelligibility assessments with or without a masker. At its simplest, SRT measurements in a sound field are done in discrete locations of signal and noise, such as conditions S0N0, S0N90 and S0N180, which are shown in Figure 4.6.

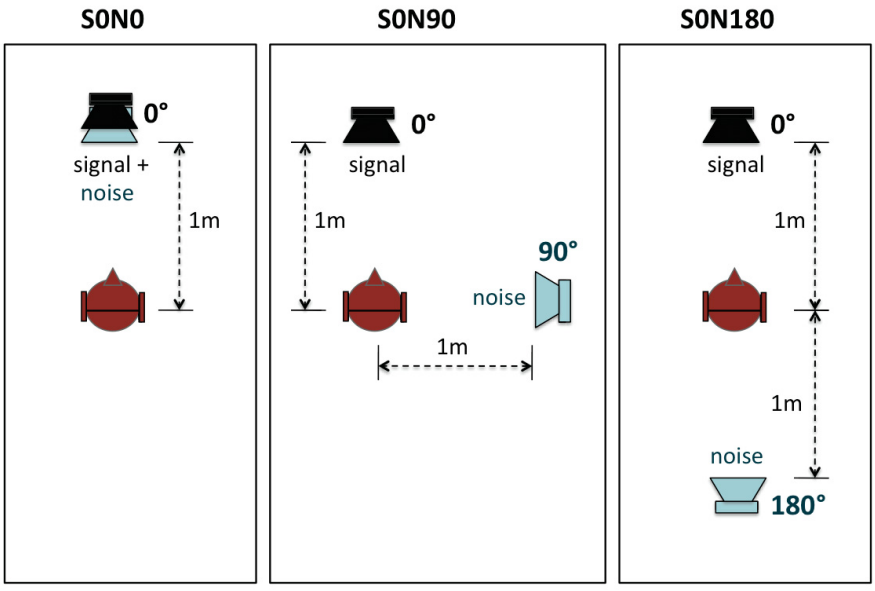


Figure 4.6: Conventional loudspeaker arrangements for testing speech intelligibility in noise in sound field. The figure is adopted from [74].

Another test to conduct in a sound field is the evaluation of the functional gain of hearing instruments. This is done for example by comparing the SRT or some other audiometric

measure with and without a hearing instrument. This allows the hearing instrument algorithm parameters to be optimized individually for the user. Also, hearing instrument manufactures are obviously interested in how their products work on individuals. For example, the performance of a directional microphone of hearing aid can be evaluated by comparing the hearing performance with different microphone settings of the same hearing aid [15]. Furthermore, testing the localization abilities is done to assess the functioning of bilateral or binaural hearing instruments [88].

Pure-tone audiometry can also be conducted in a sound field with some limitations. It has been even argued [19, 89] that the measurements in SFA should be comparable to the ones made in PTA over headphones. Opinions on this are diverse in the literature, while one of the main motivations for SFA is that it gives results which are more descriptive of the real hearing abilities compared to PTA [70]. Eventually, the audiometric test to conduct depends on the situation. Therefore, for a versatile SFA system, an option of conventional frequency specific audiometric measurements might be useful.

However, the use of pure tones in SFA is somewhat problematic. Due to acoustical room modes, there would be substantial spatial variations in the sound pressure level in the room if pure tones were used as test signals. The lack of motivation to use pure tones in SFA is stated by various authors [19, 17, 89]. Instead, warble tones [19, 89] and narrow-band noise [19] have been suggested, because they generate a more uniform sound field in the room but are still frequency specific.

4.4.3 Technical considerations

There are several downsides and technical issues in conducting audiometry in a sound field. First, the reflections, reverberation, and standing waves in the room have an effect on the sound field produced by the loudspeakers [17]. The effect of these can be minimized by increasing the absorption in the room and compensating for the room modes. An ideal test room would be completely anechoic to avoid these problems [33]. However, this is often not practical in clinical environments. Still, for the results to be comparable between different clinics, the room characteristics should be at least standardized so that the test room would have the same effect on the results everywhere. Second, any head movement of the patient affects the position of the loudspeakers relative to the patient and thus affects the results [17]. Finally, in sound-field measurements, binaural hearing is measured, where the more sensitive ear dominates. If ears are wanted to be tested separately, one ear must be occluded or masked by noise [33].

The standard ISO 8253-2 [33] defines the procedures, test tones and sound field conditions for SFA. However, only pure tones, warble tones and narrow band noise are considered as test signals in the standard. ISO 8253-2 gives guidelines for "quasi-free sound field conditions", in which the room effect is negligible in some frequency range, but which should be suitable for a relatively simple setup. The requirements for the quasi-free conditions are given in [33] as follows.

- Loudspeakers are placed at the ear height at a distance of at least 1 m from the head of the listener (reference point).
- In the left-right and up-down axis, the SPL produced by the loudspeakers at positions 0.15 m from the reference point should not deviate more than ± 2 dB (with the

listener and chair absent).

- In the front-back axis, the SPL produced by the loudspeakers at positions ±0.10 m from the reference point should not deviate more than ±1 dB from the theoretical value given by the inverse sound pressure distance law (with the listener and chair absent).

In practice, there is a low-frequency limit, below which the conditions will not be met due to room modes. Thus, it must be made sure that the frequency range of the test signals are in the range of where quasi-free field conditions are valid. This might lead to a need to high-pass filter of the test tones.

The importance of the degree of correlation of the loudspeakers used in multichannel setups was pointed out in [70, 71]. Accordingly, when correlated noise is reproduced from all loudspeakers of a symmetrical setup, the auditory event is localized inside the head, which is not generally wanted in this application. Uncorrelated maskers are instead discrete and externalized. However, too discrete sources can disable smooth auditory atmosphere. Hence, "a degree of controlled correlation" was suggested in [71] for seamless perception. In [70] in was noted, that using uncorrelated synthetic noise are not representative to real sound environments, while most real-life sound fields are constructed of both direct and diffuse sound. Indeed, when using real recorded environmental noise, a realistic amount of correlation is achieved, making the correlation a non-issue.

Another issue is the adequate number of loudspeakers. For example, assessing speech intelligibility in noise with a two-loudspeaker setup as in Figure 4.6 has some limitations. In [72] it was argued that audiometric testing with single narrow masking noise source is irrelevant, while it does not reflect real-life performance. Indeed, real-life noise is not often at distinct location but surrounds the listener. In the case of directional hearing instrument microphone, a measurement where the masker is presented in one direction is problematic. Namely, results may vary significantly depending on whether the masker is directed on the beam or blind spot of the microphone [71]. Hence, several loudspeakers are needed to enable multiple test signal locations and an enveloping masker. The insufficiency of two-loudspeaker SFA systems was concluded also in [15]. In that study, significantly different SRT scores were achieved with two-loudspeaker SFA setups compared to real situation [15]. Furthermore, in [71] it was showed that errors from head rotation and microphone directivity decrease substantially when the number of loudspeakers is increased from four to eight. Because of these facts, speech intelligibility in noise has been suggested to be assessed with larger loudspeaker setups, examples of which are given in the next subsection.

4.4.4 Sound field audiometer implementations

A variety of sound-field audiometry systems are described in the literature. Some of them are complex and can be used also in applications other than audiometry. Some systems in turn focus more on their availability for clinical environments rather than seeking for perfect reproduction.

An example of the more complex SFA systems, the "Simulated Open-Field Environment (SOFE)" setup was introduced in [79]. It consisted of 48 loudspeakers and a visual display in an anechoic room. The SOFE setup is shown in Figure 4.7. The setup was intended to be used for instance in comparing the hearing performance of normally-hearing and

hearing-impaired individuals in the same environment. A system similar to SOFE, namely the "Loudspeaker-Based Room Auralization (LoRA)" system, was introduced in [24].

Another rather large SFA implementation, namely the "Virtual Sound Environment (VSE)" system is shown in Figure 4.8. The setup consisted of 29 loudspeakers on a sphere surrounding the listener. The setup was built in a studio room, which has the reverberation time of 0.35 s below 500 Hz and 0.2 s above 500 Hz. The VSE setup has been used to compare hearing abilities of normally-hearing and hearing-impaired individuals in various acoustical environments. [54]

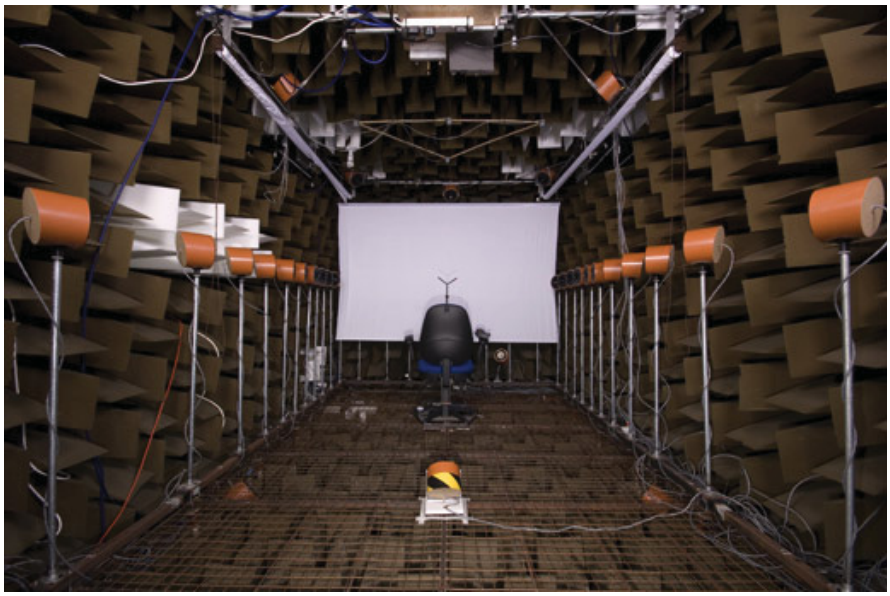


Figure 4.7: A system for sound-field audiometry called Simulated Open-Field Environment (SOFE) [79]. The figure is adopted from [78].

The idea of VSE is based on simulating a virtual room. Virtual sound sources are placed on the virtual room and room impulse responses (RIR) are calculated from the sources to the listening position. Thus, one RIR is the transfer function of one virtual loudspeaker to the listening position in that virtual room. Audio material for the audiometric test is then filtered with the RIRs and reproduced with ambisonics. This method allows for example assessing the effect of room acoustics on speech intelligibility. [54]

Although the SOFE and VSE systems provide precise simulation, they both are quite massive for clinical environments. A more simple approach was proposed in [87], namely the use of a conventional 5.1 or 7.1 loudspeaker setup. This was reasoned due to the good availability of commercial equipment and sound material for them.

A relatively simple localization test setup aimed for clinical use was proposed in [88]. In that setup, there was 13 loudspeakers placed at ear level in the frontal horizontal plane, symmetrically between -90° and 90° in 15° steps. Low-pass and high-pass noise was used as a test signal, in addition to a telephone ringing sound, which was considered a broad-band tone. To enable the same test to be done with headphones, HRTFs were measured with an artificial head from each loudspeaker. HRTFs were also measured with an artificial head wearing a behind-the-ear hearing aid. In the report, localization abilities were however

Figure 4.8: A system for sound-field audiometry called Virtual Sound Environment (VSE) [54]. The figure is adopted from [54].

shown to be somewhat poorer in headphone listening, because the HRTFs of the artificial head do not represent individual HRTFs. [88]

Another relatively simple system, namely R-SPACE, was introduced in [71]. R-SPACE is a recording and reproduction method for a loudspeaker setup with eight loudspeakers equally placed in the horizontal plane. Recording is done with a respectively-placed microphone-array of eight shotgun-type microphones. The system is especially intended to be used in testing and comparing of different hearing instruments and their features in realistic sound scenes, for example in a cafeteria ambience. The idea is that some sound scene is first recorded and then reproduced in the loudspeaker setup. When reproducing, measurements are done with an artificial head equipped with hearing instruments. In the actual test, these recording are then listened with insert headphones by normally-hearing listeners. [71]

Chapter 5

DirAC-based sound-field audiometry

Section 3.2 introduced a spatial audio technique called Directional Audio Coding (DirAC) and Section 4.4 explored the concept of sound-field audiometry (SFA). In the following chapter, a concept of sound-field audiometry system utilizing DirAC is described and motivated. Also prototyping and brief technical validation of the concept is reported.

5.1 The overall concept

The fundamental idea in DirAC-based sound-field audiometry is to enable hearing performance assessments in reproduced sound scenes with reasonable technical requirements. Using the term "reproduced sound scenes" emphasizes the fact that nothing is simulated, but existing sound environments are recorded and reproduced. Thereby, realism is achieved naturally. To be precise, however, sound scene reproduction is mixed with augmentation of external sound events.

The overall concept of DirAC-based sound-field audiometry is illustrated in Figure 5.1. The black box in the figure represents the SFA system itself, the contents of which are described in the next section. To reproduce any given sound scene, the system needs two inputs from the real environment to be reproduced. First, the background noise is recorded in A-format. Second, the acoustical properties of the environment are captured in an A-format spatial impulse response. The third input is an anechoic speech corpus. The talker of this corpus is then augmented to the sound scene using the measured acoustical properties of that environment. The system outputs the loudspeaker signals for arbitrary number of loudspeakers in arbitrary locations.

When the sound scene with the augmented talker is reproduced in the SFA setup, the listener is not only enveloped by the sound scene of the real environment, but also hears the talker in the speech corpus as if he/she was also there in the real environment. Thereby, speech intelligibility assessments, for example a SRT test in noise, can be made in realistic environments, using the background sound scene as the masker and the augmented talker as the test speech. Now that the anechoic speech corpus and environment recordings are separated, different speech materials can be flexibly combined with different background

scenes. This is advantageous, since the speech corpus itself does not have to be recorded in the real environment. That is, the system is compatible with any existing speech corpus. Also, sound scenes can be recorded anywhere with relatively simple methods. Consequently, speech intelligibility assessments can be virtually done, for example, in a highly-reverberant railway station lobby, or some specific cafeteria, depending on what kind of scenarios are of interest. Although the discussion in this thesis is limited to speech intelligibility assessments in noise, the presented concept is not necessarily limited to them.

The presented concept is motivated mainly by the limitations of traditional audiometric methods in measuring real-life hearing performance. With this concept, the patient, although physically in the clinic, can be virtually transported to the sound scene of a real environment of choice. Another advantage in the concept is that loudspeaker signals are naturally uncorrelated. On the contrary, if an omnidirectional recording of the background noise was reproduced with several loudspeakers, the loudspeaker channels would correlate.

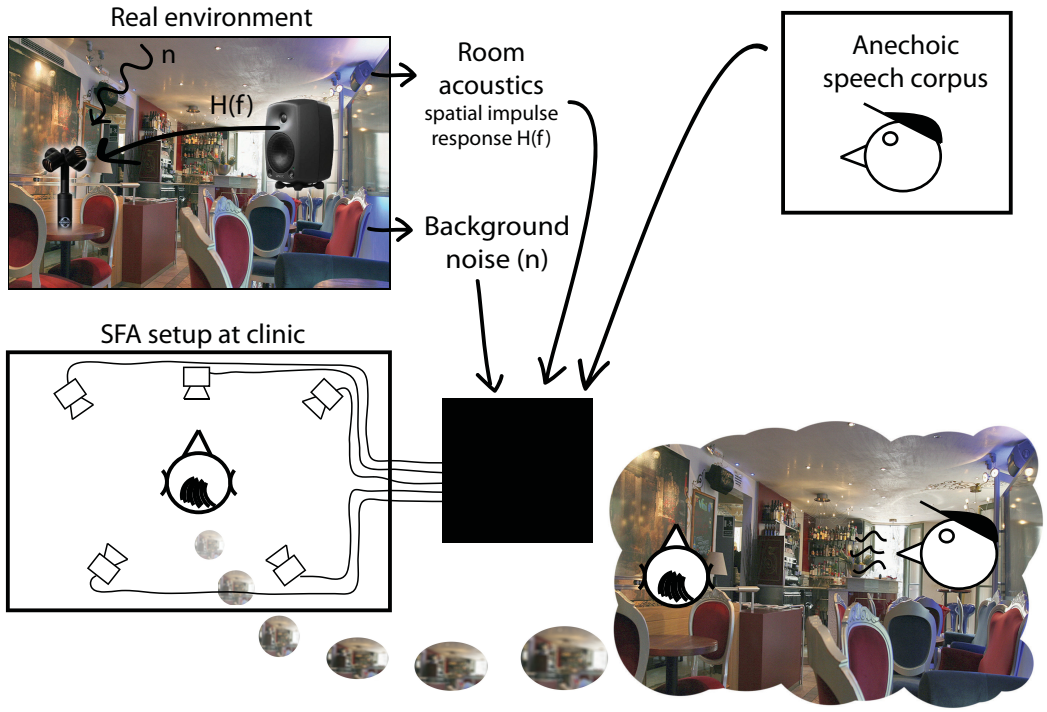


Figure 5.1: The overall concept of DirAC-based sound-field audiometry.

5.2 Description of the SFA system

5.2.1 Reproducing a sound scene

The black box in Figure 5.1 represents the SFA system itself, consisting of the chain of operations from the original recorded audio material to the final loudspeaker signals for the test setup. A high-level block diagram of this black box is presented in Figure 5.2. The signal flow consists of two separate audio streams, namely the masker stream (from

input 1) and test speech stream (from inputs 2 and 3) which are combined not until in the audio interface.

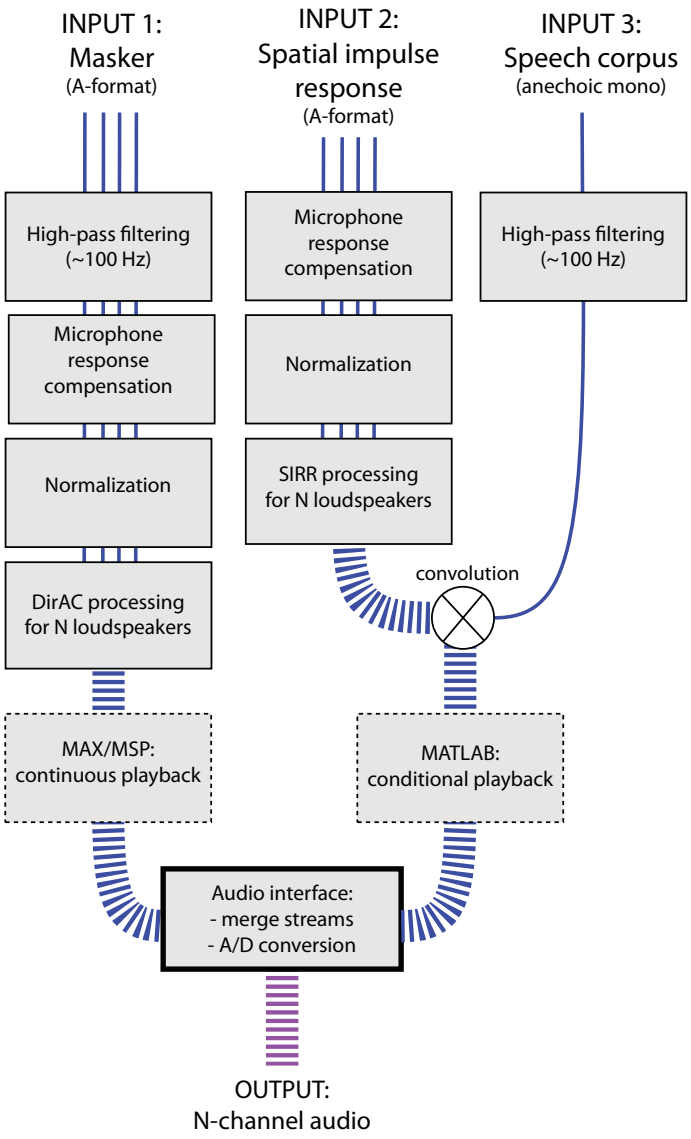


Figure 5.2: Block diagram of the DirAC-based SFA system.

In the beginning of the masker stream, the four-channel A-format background noise file is high-pass filtered to avoid problems with differing rooms modes in different test rooms. The cut-off frequency should be decided depending on the Schroeder frequency of the test room. In the tests of this thesis, a cut-off frequency of 100 Hz was found to be suitable. Next, each of the microphone capsule signals are filtered with a compensation filter, calculated as the inverse of the SPS200 A-format microphone on-axis capsule responses (see appendix A for details). Then, the file is normalized to compensate for different signal levels in different original recordings. Finally, the four-channel file is sent to DirAC-processing block, in which parameters are given, such as the number of reproduction

loudspeakers (N). This block outputs a N-channel audio signal, with one channel for each loudspeaker. This file is in continuous playback via MAX/MSP software and is unaffected by the processing in the second stream.

The second steam (i.e., the test speech stream) is formed from the A-format spatial impulse response and the single-channel anechoic speech corpus word lists. First, identically to the masker, the spatial impulse response goes through microphone response compensation and normalization. Then the four-channel file is sent to SIRR-processing block, in which respective parameters are used as in the DirAC-processing block of the masker. The SIRR-processing block outputs the N-channel impulse response, one channel for each loudspeaker. This N-channel impulse response is convolved with the high-pass filtered test speech. This results to N-channel test speech audio signal, with one channel for each loudspeaker. These files are then handled by the test logic in MATLAB software. The test logic analyzes the word locations in the test speech files and runs the actual test.

For testing the concept, a SRT-test logic was implemented in MATLAB which plays back the files and changes the SNR depending on the patient's answers. The core of the test logic was based on the code used in [2], however with extensive modifications. The test logic used the PlayRec script [29] for the audio playback.

5.2.2 Usage of the system in the test conductors viewpoint

Usage of the DirAC-based sound-field audiometry system is straightforward. A screen capture of the test conductor interface is presented in Figure 5.3.

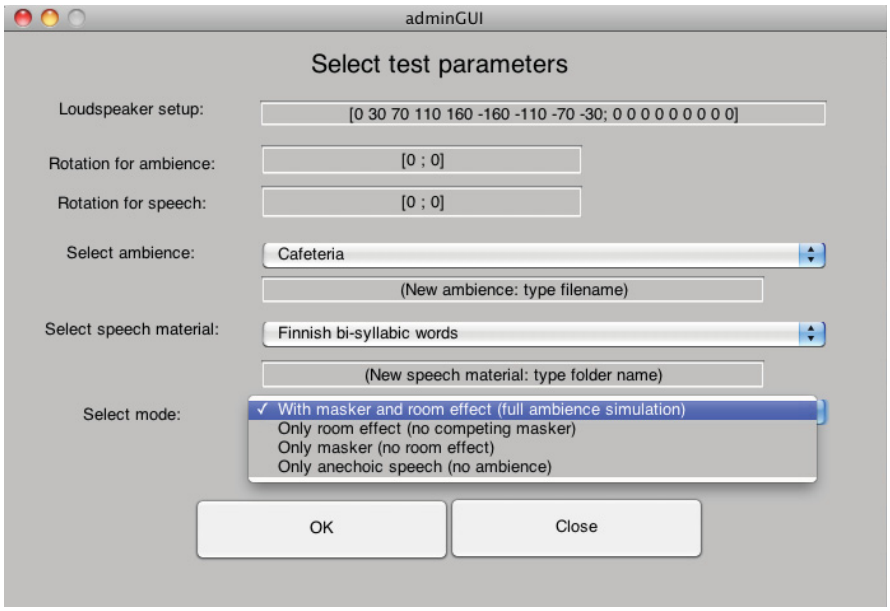


Figure 5.3: The DirAC-based SFA system user interface for the test conductor. Test parameters are selected in the beginning of each test with this interface.

First, the locations of the loudspeaker are defined in azimuth and elevation. Then, the desired combination of sound scene and test speech material is selected. Additionally, the masker stream and speech stream can be rotated or tilted, allowing for example to change

the talker location. It is also possible to conduct the test without a masker or without a room effect. In the latter case, anechoic speech is used. After setting the test parameters, an interface for the listener opens up, including a bare text field for the answers and the test can begin. Another figure opens up for the test conductor where the adaptive track can be monitored.

As all the three inputs are separated, combinations can be done flexibly. That is, it is possible to combine a masker from one real environment to room acoustics from another environment. This was not however implemented in the interface shown in the figure, because the system was wanted to keep as simple and usable as possible. Anyhow, one of the advantages of the software is its modularity. Namely, if new inputs are wanted to be used, they can be flexibly added to the system by just typing a folder name where the audio files are located. Thus, the end user is not restricted to use the supported sound scenes, but is able to import new test speech material or even own A-format recordings.

5.3 Prototyping

5.3.1 On the test environments

For prototyping of the concept, three test environments were built. First of all, a reference environment was built to represent a real-life situation. Then, two SFA prototype setups were built. First of the prototypes represents clinical setup, a setup compact enough that could be implemented in an audiometric clinic. The second prototype represents an ideal setup in anechoic conditions, a setup free from the additional room effect of the test room. The sound scene in the reference environment was captured and then reproduced in the two prototype setups following the methods introduced previously in this chapter. The environments were arranged so that the same SRT test could be done in all three environments. In the next three subsections, the test environments are described.

5.3.2 Reference environment

The reference environment was designed to represent a realistic environment with background noise and reverberation, such as a crowded cafeteria. Real environments usually have a non-stationary background noise, but using such would induce reliability issues to the actual audiometric test. Thus, a conscious compromise between realism and test reliability was made: the background noise was restricted to be rather stationary.

The reference environment was built in a relatively big room equipped with eight active loudspeakers (Genelec 1030A) placed fairly uniformly around the room. Each of these loudspeakers were tilted differently in elevation and directed towards wall, bookshelf, or other reflective and diffusive surface. This was done to avoid direct sound from these loudspeakers to the listener position and to make the sound field as diffuse and enveloping as possible. Anechoic 18-talker speech babble was played back from these eight loudspeakers as the background noise. The babble was mixed from anechoic recordings of nine different male talkers. The recorded material consisted of complete sentences in Finnish. Due to that many overlapping talkers, the informational content in the sentences could not be

followed. The babble was mixed so that the sentences of different talkers were overlapping, resulting in quite uniform sound pressure level in time. Individual talkers were also balanced in SPL. Additionally, even though the same babble was used in all loudspeakers, the files were unsynchronized for 2–15 seconds between the loudspeaker channels, resulting in incoherent playback. That is, the same part of the babble file never occurred simultaneously in more than one loudspeakers.

The speech babble files were finally high-pass-filtered with a 100 Hz cut-off frequency. This cut-off frequency was chosen while it roughly matched the Schroeder-frequencies in the reference environment room and in the listening room (discussed in the next subsection). Normally high-pass filtering would happen after recording the audio material, as was shown in Figure 5.2. Now the filtering was done here to maintain the comparability of the reference environment and the prototype setups.

In addition to the eight loudspeakers providing the background noise, one active loudspeaker (Genelec 8030A) was used to produce the test speech. This loudspeaker was placed in 2 m distance from the listening position and directed towards it. A distance that high was chosen to decrease the the direct-to-reverberation ratio in the listener position. This made sure that when anechoic speech material was played back from this loudspeaker, the room reverberation was clearly audible in the listener position. The height of the loudspeaker was 1.2 m, which was the average ear-height in this situation. All audio in the reference environment was controlled with a computer (Apple Mac Pro) connected to an audio interface (MOTU UltraLite-mk3 Hybrid). The background noise was played with MAX/MSP and the test speech from the SRT-test software in MATLAB.

In the reference environment room, all walls, floor, and ceiling were concrete but there was a good amount of miscellaneous more and less absorptive material. To get more insight of the acoustical properties of the room, the reverberation time was measured. The measurement procedures are described in appendix B. The measured values of RT are presented in Table 5.1. Table 5.2 presents the room dimensions and the Schroeder frequency calculated with Equation 2.5, using the average value of RT as an input.

Table 5.1: Reverberation time (RT) and dimensions of the reference environment.

Octave band [Hz]	63	125	250	500	1000	2000	4000	8000	average
RT [s]	0.69	0.54	0.46	0.43	0.45	0.44	0.39	0.31	0.44

Table 5.2: Dimensions and Schroeder frequency of the reference environment.

Length	8.7 m
Width	6.2 m
Height	3.6 m
Volume	194 m^3
Schroeder frequency	95 Hz

As the reference environment was to represent the "real environment" in Figure 5.1, the background noise and room acoustics were captured as follows. The speech babble was calibrated to level of 65 dBA SPL and a one-minute sample was recorded with an A-format microphone (Soundfield SPS200) placed in the listening position. A spatial impulse response was measured from the Genelec 8030A loudspeaker used for test speech reproduc-

tion to the Soundfield SPS200 microphone in the listening position. The measurement was done using a logarithmic sine sweep and post-processing procedures similar to which were explained in [52], a report describing similar measurements done in a concert hall. All measurements were done in MATLAB environment with a computer (Apple Macbook) connected to an audio interface (MOTU Traveler mk3).

5.3.3 The listening room prototype setup

The first prototype for the DirAC-based SFA system was built in a listening room that meets the ITU-R BS.1116 [35] listening room recommendations. This setup was to represent a clinical setup in terms of the room size and acoustics. The room was equipped with nine active loudspeakers (Genelec 8260A) in floor stands and four active loudspeakers (Genelec 8240A) attached to the ceiling. The number of loudspeakers used and their location was varied during the validation process. All audio was played back with a computer (Apple Macbook) connected to a digital mixer (Yamaha 02R96) via an audio interface (RME Fireface 800).

Reverberation time in the listening room was measured using measurement procedures described in appendix B. The measured values of RT are presented in Table 5.3. Table 5.4 presents the room dimensions and the Schroeder frequency calculated with Equation 2.5, using the average value of RT as an input.

Table 5.3: Reverberation time (RT) and dimensions of the listening room prototype setup.

Octave band [Hz]	63	125	250	500	1000	2000	4000	8000	average
RT [s]	0.42	0.38	0.23	0.25	0.26	0.27	0.26	0.23	0.26

Table 5.4: Dimensions and Schroeder frequency of the listening room prototype setup.

Length	6.3 m
Width	5.6 m
Height	2.7 m
Volume	95 m^3
Schroeder frequency	105 Hz

5.3.4 The anechoic prototype setup

The second prototype for the DirAC-based sound-field audiometry system was built in an anechoic chamber. This setup was to represent the ideal setup in terms of room acoustics. The chamber was equipped with active loudspeakers (Genelec 8030A) in a transformable 3D-setup. All audio was played back with a computer (Apple Mac Pro) connected to an audio interface (MOTU 2408 mk3).

5.4 Technical validation

5.4.1 Sources of error in the reproduction chain

Figure 5.4 summarizes the sources of error in the path from the reference environment to the prototype setups. In other words, that is the simplified transfer function from the sound event in the reference environment to the sound event in the SFA prototype setup room. First error is due to unideal microphone response. The compensating filter corrects most of this error, but does not account for the high-frequency roll-off above 18 kHz[1]. The next error is due to the possible artifacts from the DirAC-processing. Finally, there are the loudspeaker responses and the listening room response in the reproduction chain.

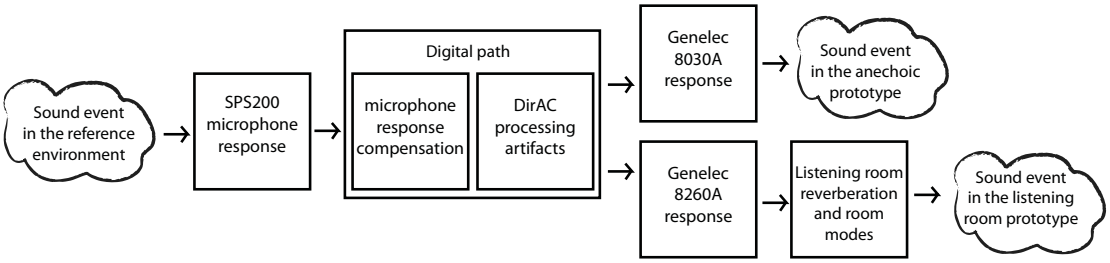


Figure 5.4: Sources of error in the SFA system reproduction chain.

The figure shows, that by comparing the sound events in the three test environments, information is gained about how much error is introduced to the reproduction by the processing chain of the SFA system, and on the other hand, how much by the semi-reverberant test room. When comparing the sound events, it should be noted that different loudspeakers types were used in the two prototype setups.

To define the effect of the error sources, two brief measurements were done. First, the sound event produced by the masker was recorded in all three test environments with a measurement microphone. The magnitude spectrums of these recordings were compared to evaluate how much spectral coloration there is in the prototype reproduction setups compared to the reference environment. Second, the sound event separately produced by the masker and a test speech word list was recorded with an artificial head in the three test environments. The timbre and amount of reverberation of these recordings were subjectively analyzed. For these measurements, nine loudspeakers were used in both prototype setups and they were arranged in azimuths 0°, ±30°, ±70°, ±110°, and ±160°, all in the horizontal plane. In the listening room prototype setup, loudspeakers were located 2.3 meters from the microphone.

5.4.2 Magnitude spectrum comparison

In the first measurement, a one-minute sample of the masker was recorded in the three environments. Measurements were done with a microphone (B&K, type 4192 capsule and type 2669 preamplifier) and a conditioning amplifier (Nexus) connected to a computer (Apple Macbook) via an audio interface (MOTU Traveler mk3). The long-term average

[1]The correction filter response is presented in appendix A.

spectrum was computed in MATLAB. For additional reference, the figure presents the long-term average spectrum of a word list from a speech corpus developed in [37] and widely used in speech audiometry. The list consisted of 25 bisyllabic anechoic words, talked in Finnish by a female talker.

The magnitude spectrums are shown in Figure 5.5. The figure shows that the spectral envelope is of similar shape in all three cases. However, there are fluctuations up to 4 dB in magnitude thorough the audible range. The fluctuations could be explained most likely by the loudspeaker response in the prototype setups and the room modes in the listening room. The test speech spectrum has similar spectral envelope than the maskers. This means that when using this masker and this test speech corpus, the SNR is quite constant across the frequency range. However, the fundamental tone (and it's harmonics) of the test speech is higher, because the talkers in the masker are male and the test speech is talked by a female.

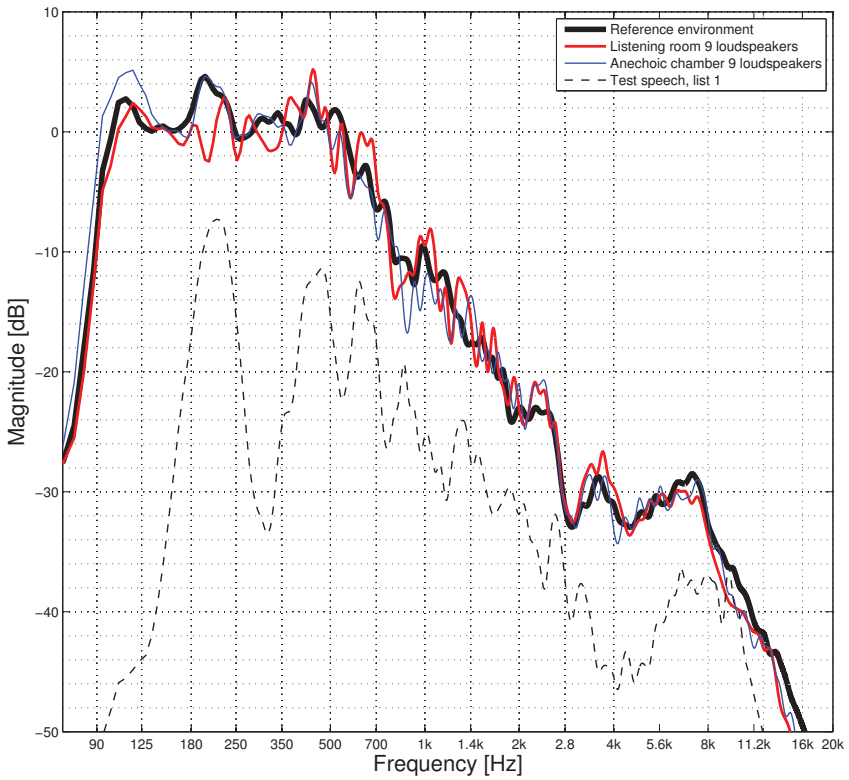


Figure 5.5: A comparison of the magnitude spectrums of maskers played back in the test environments (the three upper curves) and a word list file from the speech corpus described in [37] (the lower curve).

5.4.3 Informal observations from binaural recordings

The magnitude spectrum comparison shows some difference in the spectrums but does not clarify whether it is negligible or not. It is difficult to subjectively evaluate the reproduction quality in the test environments, as they are physically in different rooms. Thus, binaural

recordings of the masker and test speech were made in all the three test environments. The same masker and speech corpus word list that was used in the magnitude spectrum comparison were played back and recorded in the three environments. For the test speech this means that in the reference environment the original anechoic word list was used and in the prototype setups the SIRR-processed list was used – just as it is done when a SRT test in the system is conducted. These measurements were done with an artificial head (Cortex Manikin MK1) connected to a computer (Apple Macbook) via an audio interface (MOTU Traveler mk3).

The binaural recordings were carefully listened with headphones (Sennheiser HD800) by the author. The recordings from the prototype setups were compared to the ones from the reference environment. Ideally, there should be no audible difference between these. Observations were made as follows, concerning spectral coloration and possible excess reverberation in the reproduction.

First of all, coloration was slightly audible in the maskers in both prototype setups. On the other hand, in the maskers there were no significant coloration audible and especially the speech recordings from the anechoic prototype and the reference environment was really hard to tell apart.

Secondly, there was some excess room effect audible in the speech recording from the listening room prototype. This is obvious, while there is an extra room response involved, namely the physical one of the listening room. However, this was not audible in the continuous masker recordings.

Finally, there was no excessive room effect in the recordings from the anechoic prototype. This was a clear improvement compared to what was achieved with B-format DirAC-processing. Namely, the same binaural measurements were conducted earlier by the author using DirAC-processing with B-format input signals. In those recordings, a clearly audible increase in the room effect was noticed, initially motivating the use of A-format DirAC-processing in this application. This finding gives further evidence for the note in [61]: compared to B-format processing, A-format DirAC-processing is more precise with the diffuseness estimate in high frequencies. Indeed, now with A-format DirAC-processing, there was no audible increase in room effect, while the diffuseness was not overestimated.

5.4.4 Conclusions

Based on the magnitude spectrum comparison, there were some differences in the spectral content of the maskers in the SFA prototype setups compared to reference environment. Also, this difference could be subjectively noticed by listening recordings of the respective maskers. In the case of test speech, the subjective difference was negligible. These findings emphasize the complexity of the sound scene reproduction system: there are many cascaded factors affecting to the final sound event. The technical validation suggested that the system could be valid, but did not prove it. This gives motivation for further validation with listening tests, which are reported in the next chapter.

Chapter 6

Subjective listening tests

The psychoacoustic listening tests reported in this chapter were made to further evaluate whether the concept of DirAC-based sound-field audiometry is valid. The concept is considered valid, if equal scores of some audiometric measure can be recorded in a real environment and in a DirAC-based SFA setup in which the first mentioned is reproduced. If this criterion is met, hearing diagnosis made in a DirAC-based SFA setup can be considered to represent real-life hearing performance in terms of that particular audiometric measure. Furthermore, the listening tests are aimed to suggest the adequate number of loudspeakers and the requirements for the test room acoustics, with which the validity criterion is met.

6.1 Test method in general

To test the equal speech intelligibility between the reference of real environment and its DirAC-reproduced equivalent, a set of listening test was conducted, where the test subject went though a SRT-test in a real environment and corresponding reproduction environments. The reference environment described in Section 5.3.2 was used as the real environment and the prototype setups described in Sections 5.3.3 and 5.3.4 were used as the reproduction environments. Test procedures were the same as if speech audiometry with SRT-procedure was conducted. Thus, the validity criterion is met if the test environment has no effect on the SRT. This is proved if there is no statistically significant difference between the SRTs recorded in the different environments and the data used in the analysis has confidence intervals not larger than what would be accepted as an error in clinical SRT-measurement.

Total of three separate listening tests were made and they are reported in Sections 6.2–6.4. In all three experiments, a SRT test procedure was used with the same speech material. Test A was somewhat a preliminary experiment for piloting the concept. Based on the information gained from that experiment, the test arrangement was refined for tests B and C. In test B, normal hearing test subjects were used and in test C the procedure was repeated with hearing-impaired test subjects, who all had either hearing aid, cochlear implant, or both in use. In all three tests, the general test method was similar to allow comparison. Figure 6.1 summarizes the listening test setups.

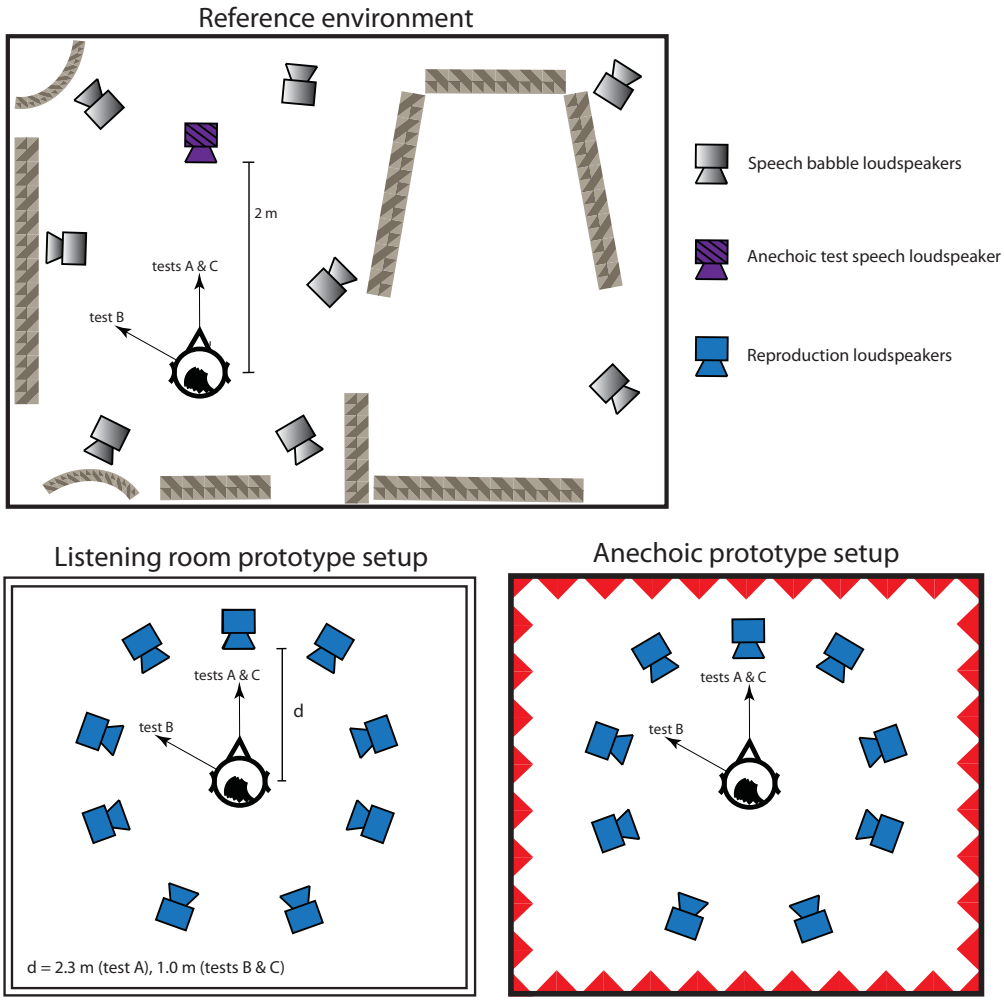


Figure 6.1: The listening test setups. The arrows indicate the test subject orientation.

6.2 Test A

6.2.1 Introduction

For each test subject, SRT was measured in the following scenarios.

1. MON* - Mono reproduction in the listening room prototype with one loudspeaker
2. LR5* - DirAC-reproduction in the listening room prototype with five loudspeakers
3. LR9* - DirAC-reproduction in the listening room prototype with nine loudspeakers
4. LR13* - DirAC-reproduction in the listening room prototype with 13 loudspeakers
5. LRA* - Anechoic test speech and DirAC-reproduced masker in the listening room prototype with 13 loudspeakers
6. REF - The reference environment

The scenarios 2, 3, and 4 were identical except for the number of loudspeakers. In the LR5*-scenario, five loudspeakers were arranged in a standard 5.1-setup (without separate subwoofer), in azimuths 0°, ±30°, and ±110° in the horizontal plane. The LR9*-scenario was identical with the latter but with extra loudspeakers in the azimuths ±70° and ±160° in the horizontal plane. Finally, the LR13*-scenario was identical to LR9*, but with four extra loudspeakers with azimuth ±45°, and ±135°, all elevated by 45°. The scenario LRA* was identical to LR13* except that anechoic speech material was used.

The MONO-scenario was used as the reference for the most simple SFA system. Here, both the masker and the test speech were reproduced from the front loudspeaker (i.e., azimuth 0°, elevation 0°). The mono-version of the masker was generated by summing the four A-format channels from the original masker recording done in the reference environment. Summing the channels resulted into an omnidirectional response. Similarly, the mono-version of the impulse response was done by summing the channels of the original A-format spatial impulse response.

In the listening room, for scenarios 1–5, all loudspeakers were positioned 2.3 m from the listening position. The loudspeakers were calibrated to that positioning with the dedicated calibration software by the loudspeaker manufacturer. In the reference environment, the listening position was the same as the microphone position when capturing the sound scene (discussed in Section 5.3.2). In all scenarios, the test subjects were positioned facing forward, that is, facing the front loudspeaker in the listening room and facing the test speech loudspeaker in the reference environment.

6.2.2 Test subjects

Twenty-one test subjects with normal hearing thresholds participated in test A. The test subjects, 18 of which males and three females, were all native Finnish speakers (age between 23–42). The test subjects were volunteers and were not given any reward for participating in the test.

6.2.3 Test procedures

An adaptive 1-up/1-down method [48] was used to measure the SRT in noise. This was done separately for the six scenarios. Each test subject went through every scenario. The speech material used was a Finnish speech corpus [37] of six word lists, each with 25 phonetically balanced bisyllabic words, talked by a female talker[1]. For each scenario / adaptive track, one word list was used. That is, test subjects heard each of the 150 words only once. All 25 words were played regardless of the reversals in the track and the SRT for each track was calculated as the average of the six last reversals.

The masker level was kept constantly at 65 dBA SPL and the test speech level was variated depending on the answers. Initial step size for the adaptive track was 6 dB and it was decreased to 2 dB after two reversals in the track. Initial SNR of the track was 6 dB for all scenarios except scenario 1, in which the initial SNR was 3 dB. The SNR was limited between –60 and +12 dB. The combinations of which list was used in which scenario for

[1]This corpus was also utilized in the technical validation, reported in section 5.4.

which subject was governed by a truncated latin square. This ensured that each list was used equally within scenarios. The scenarios were conducted in a fixed order: 5–1–2–3–4–6.

Test subjects were first introduced to the test procedure with written and spoken instructions. Then, before the actual test began, a training sessions was held to familiarize the test subjects to the test procedures. In the training, a different word list was used.

6.2.4 Results and analysis

The results are shown in Figures 6.2 and 6.3. Figure 6.2 shows the individual SRT-scores in the six scenarios for all 21 test subjects. Figure 6.3 shows the marginal means and 95% confidence intervals of the SRT scores in the six scenarios.

Figure 6.3 shows quite clearly that the scenario had an effect on the SRT. To ascertain this, the results were analyzed with one-way analysis of variance (ANOVA), where the scenario was modeled as a fixed variable and the test subject as a random variable. Table 6.1 shows the ANOVA output and confirms that the scenario has a significant effect to the SRT ($p \ll 0.05$).

Table 6.1: ANOVA results for test A.

	Df	Dfd	Sum Sq	Mean Sq	F value	p
Scenario	5.00	100.00	385.14	77.03	23.38	0.00

Based on a visual inspection of Figure 6.3, it seems that the number of loudspeakers in the DirAC reproduction scenarios does not affect to the SRT by much. In the monophonic reproduction, the mean SRT is somewhat higher than in the DirAC-reproduction, although only 2 dB at most. There is a difference of approximately 3 dB between the reference and the DirAC reproduction scenarios. The only scenario that seems not to be significantly different from the reference is LRA*, in which the test speech was anechoic.

To gain deeper insight of the differences of the scenarios, a post-hoc analysis was made using Dunnett's modified Tukey-Kramer pairwise multiple comparison test [20]. The output from the post-hoc test is presented in Table C1 in appendix C.1. The post-hoc test confirms that all scenarios except LRA* are significantly different from REF. Furthermore, there is no significant difference between the DirAC-reproduction scenarios (i.e., scenarios LR5*, LR9*, LR13*). That is, the number of loudspeakers did not have significant effect on the SRT.

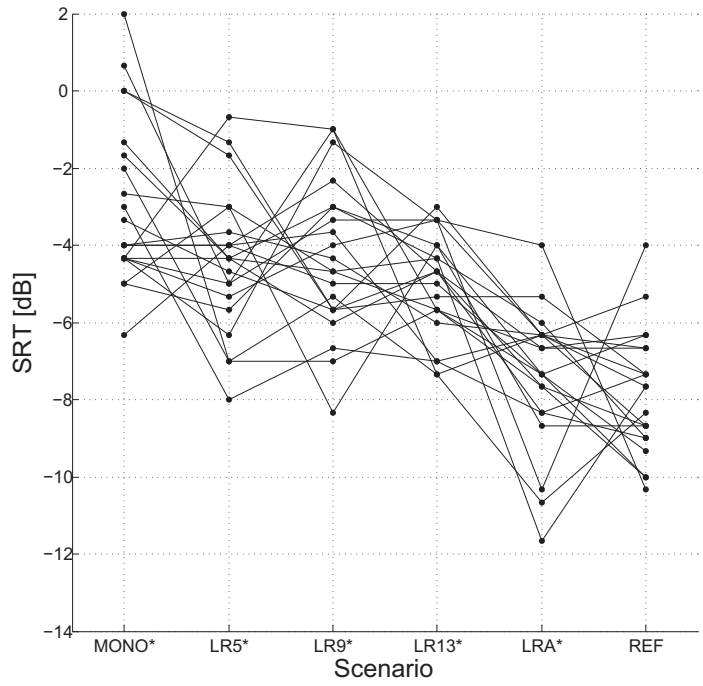


Figure 6.2: Test A results: Speech Recognition Threshold (SRT) in decibels in the six test scenarios. Individual SRT-scores of all 21 test subjects are presented. Individual scores are connected with lines for clarity.

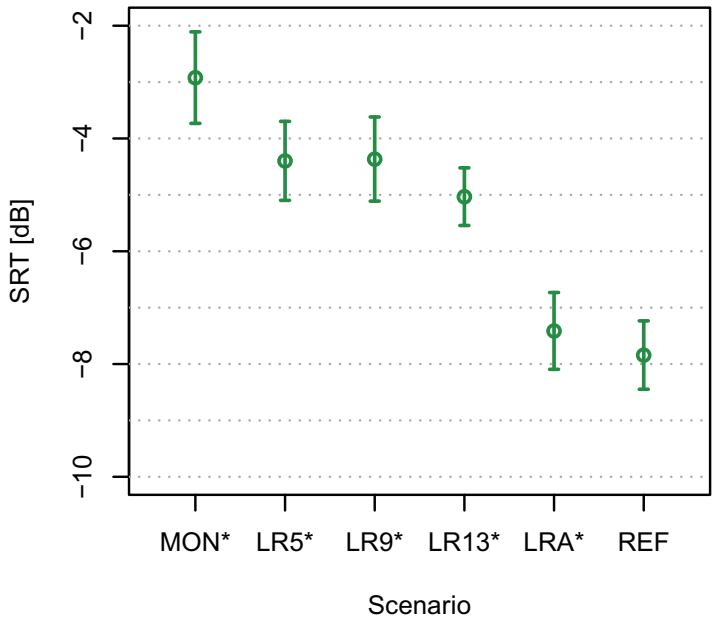


Figure 6.3: Test A results: Speech Recognition Threshold (SRT) in decibels in the six test scenarios. Marginal means of the SRT with 95% confidence intervals are presented.

6.3 Test B

6.3.1 Introduction

For each test subject, SRT was measured in the following scenarios.

1. MONO - Mono reproduction in the listening room prototype
2. LR5 - DirAC-reproduction in the listening room prototype with five loudspeakers
3. LR9 - DirAC-reproduction in the listening room prototype with nine loudspeakers
4. AC5 - DirAC-reproduction in the anechoic prototype with five loudspeakers
5. AC9 - DirAC-reproduction in the anechoic prototype with nine loudspeakers
6. REF - The reference environment

The listening room scenarios (MONO, LR5, and LR9) were similar to respective scenarios in test A (MONO*, LR5*, and LR9*), with the exception that now all the loudspeakers were positioned at a distance of 1 m from the listening position. The distance of the loudspeakers was decreased to increase the direct-to-reverberant ratio at the listening position, as this was assumed to decrease the error in SRT for these scenarios. The loudspeakers were calibrated to that positioning with the dedicated calibration software by the loudspeaker manufacturer. Scenarios AC5 and AC9 were identical to the LR5 and LR9, respectively, but conducted in the anechoic prototype setup. In these scenarios, listening position was in the middle of the loudspeaker array equidistant of the loudspeakers.

Unlike in test A, the test subjects in test B were positioned facing to azimuth -60°, that is, their right ear directed towards the front loudspeaker, which is in azimuth 0°. This was done to maximize BILD. At the presentation azimuth of ±60°, BILD is approximately 2.5 dB higher compared to presentation azimuth of 0° [8]. Thus, this positioning was assumed to give the most binaural benefit in the test and thereby be the most effective in revealing differences between the scenarios. Additionally, this positioning was assumed to maximize the test subjects' performance in the test. Namely, with uncorrelated enveloping noise, the best intelligibility is achieved when the desired sound is presented from the azimuth ±60° [8]. Indeed, people sometimes turn their head to this angle when trying to listen in interfering noise. This behavior was noticed with some test subjects in test A. Thereby, as the test subjects were now initially in the tilted position, more consistent results were assumed.

6.3.2 Test subjects

Eighteen test subjects with normal hearing thresholds participated in test B. The test subjects, 16 of which males and two females, were all native Finnish speakers (age between 23–42). The test subjects were volunteers and were not given any reward for participating in the test.

6.3.3 Test procedures

The test procedures in test B were identical with the procedures in test A with the following two exceptions. First, the initial SNR of the track was 0 dB for all scenarios. Second, the order in which the scenarios were conducted was governed with a latin square to avoid learning effects.

6.3.4 Results and analysis

The results are shown in Figures 6.4 and 6.5. Figure 6.4 shows the individual SRT-scores in the six scenarios for all 18 test subjects. Figure 6.5 shows the marginal means and 95% confidence intervals of the SRT scores in the six scenarios.

Figure 6.5 shows quite clearly that the scenario had an effect to the SRT. To ascertain this, the results were analyzed with one-way analysis of variance (ANOVA), where the scenario was modeled as a fixed variable and the test subject as a random variable. Table 6.2 shows the ANOVA output and confirms that the scenario has a significant effect to the SRT ($p \ll 0.05$).

Table 6.2: ANOVA results for test B.

	Df	Dfd	Sum Sq	Mean Sq	F value	p
Scenario	5.00	85.00	295.86	59.17	17.07	0.00

Based on visual inspection of Figure 6.5, seems that both the amount of room reverberation and the number of loudspeakers have some effect on the SRT. However, the effect is quite subtle at least in the DirAC-reproduction scenarios. In the monophonic reproduction, a bias of 5 dB can be seen compared to the reference.

To gain deeper insight of the differences of the scenarios, a post-hoc analysis was made using Dunnett's modified Tukey-Kramer pairwise multiple comparison test [20]. The output from the post-hoc test is presented in Table C2 in appendix C.2. According to the post-hoc test, the MONO-scenario is significantly different from all the others. Scenarios AC9, AC5, and LR9 have no significant difference compared to REF. Without the scenario LR5 there would be two distinct subgroups (MONO and the others), but scenario LR5 is significantly different not only from MONO, but also from AC9 and REF.

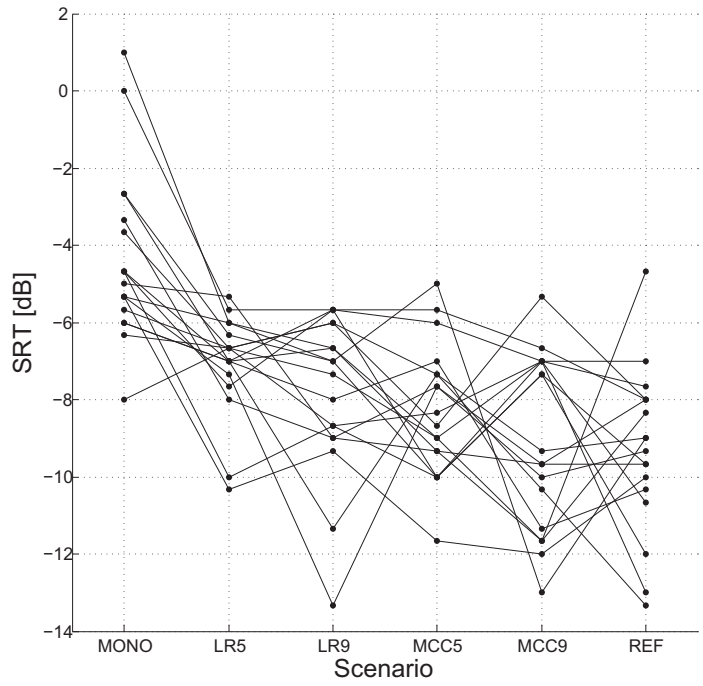


Figure 6.4: Test B results: Speech Recognition Threshold (SRT) in decibels in the six test scenarios. Individual SRT-scores of all 18 test subjects are presented. Individual scores are connected with lines for clarity.

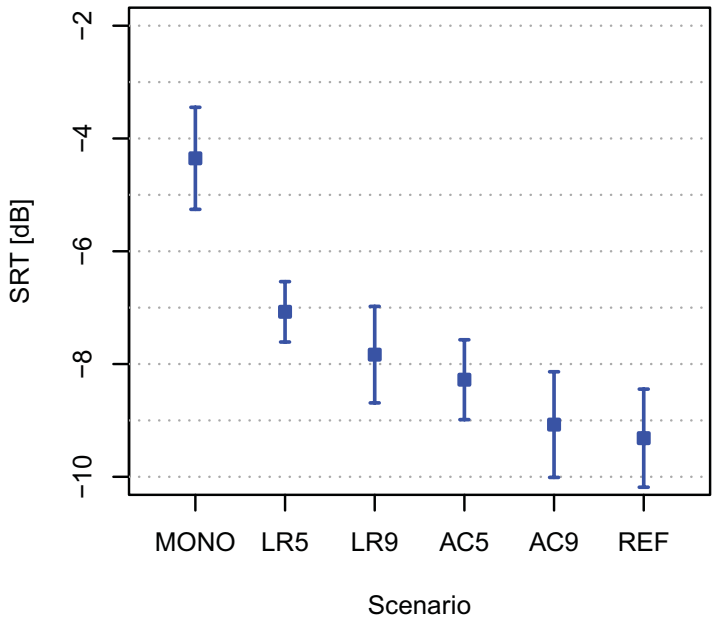


Figure 6.5: Test B results: Speech Recognition Threshold (SRT) in decibels in the six test scenarios. Marginal means of the SRT with 95% confidence intervals are presented.

6.4 Test C

6.4.1 Introduction

The aim in test C was to repeat test B using hearing instrument users as test subjects. In this test, for each subject, SRT was measured in the same six scenarios as in test B, described in Section 6.3.1. The only difference in the scenarios compared to test B was that now the test subjects were positioned facing to the front loudspeaker. This positioning was assumed to give generally the best – and also the most uniform – speech intelligibility among the test subjects in test C. The hearing-impaired test subjects were assumed to have negligible benefit from the increased BILD in the -60° position. Additionally, the tilted positioning would probably have given more benefit to test subjects with their better ear on the right. Using the the forward-facing positioning in test C was assumed to retain the comparability of tests B and C, while now both test subjects groups were positioned for the best and most uniform hearing performance in the test.

6.4.2 Test subjects

Eight test subjects participated in test C. The test subjects were all females and native Finnish speakers of the age 36 to 62. The test subjects were volunteers and they were paid an honorarium of 15 € for participating in the test. All test subjects had bilateral hearing loss which was being aided unilaterally or bilaterally by a hearing aid, cochlear implant, or both. Table 6.3 summarizes the hearing abilities of the test subjects.

Table 6.3: Test subjects in test C: hearing loss types and hearing instrument types. HA refers to hearing aid and CI to cochlear implant. Test subject RL had a wireless microphone in her almost-deaf right ear, which was connected to the aid in the left ear.

	Left ear		Right ear	
ID	Hearing loss type	Instrument	Hearing loss type	Instrument
RN	sensorineural	CI	sensorineural	HA
MH	sensorineural	CI	sensorineural	none
SH	deaf	none	sensorineural	HA
SS	sensorineural	CI	sensorineural	HA
SK	sensorineural	CI	sensorineural	HA
JV	deaf	none	sensorineural	CI
RL	sensorineural	microphone	sensorineural	HA
RP	sensorineural	none	sensorineural	CI

6.4.3 Test procedures

Test C consisted of two consecutive parts. Both parts consisted of the same SRT-procedure as did one test round in test B, with the following changes.

The first part was considered adaptation and second part was the actual test. The test subjects were not informed about this. In the first part, the masker level was 65 dB SPL.

Initial SNR of the adaptive track was 10 dB and the SNR was limited between -60 and +20 dB. During the first part, the adaptive track was carefully monitored. If it seemed that the maximum SNR of +20 dB was not adequate for the test subject, the masker level was reduced to 55 dB SPL and the initial SNR was increased to 20 dB for the second part. This procedure made sure that in the second part, as the actual data was collected, no ceiling effects were present and the masker level was at is maximum while the maximum SPL in the test room was limited to 85 dB SPL at all times.

Before the test began, the test subjects were asked to set their hearing devices to such setting that they would generally use in communication situations with background noise. They were also asked to adjust the gain of their hearing instruments to desired level during the training session.

6.4.4 Results and analysis

The results are shown in Figures 6.6 and 6.7. Figure 6.6 shows the individual SRT-scores in the six scenarios for all the eight test subjects. Figure 6.7 shows the marginal means and 95% confidence intervals of the SRT scores in the six scenarios. The masker level was 65 dB SPL for all test subjects except test subject SK, for whom the masker level was reduced to 55 dB SPL to avoid ceiling effects.

Figure 6.7 shows that although the marginal means deviate up to 4 dB between scenarios, the 95% confidence intervals are clearly overlapping. According to visual inspection of the figure, seems that the scenario does not have significant effect on the SRT. To ascertain this, the results were analyzed with one-way analysis of variance (ANOVA), where the scenario was modeled as a fixed variable and the test subject as a random variable. Table 6.4 shows the ANOVA output and confirms that the scenario had no significant effect to the SRT ($p > 0.05$).

Table 6.4: ANOVA results for test C.

	Df	Dfd	Sum Sq	Mean Sq	F value	p
Scenario	5.00	35.00	80.65	16.13	1.70	0.16

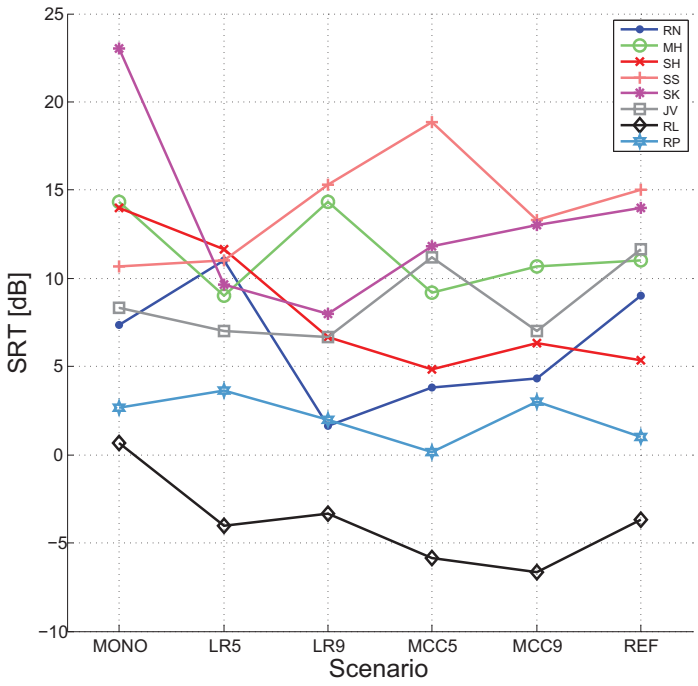


Figure 6.6: Test C results: Speech Recognition Threshold (SRT) in decibels in the six test scenarios. Individual SRT-scores of all eight test subjects are presented. Individual scores are connected with lines for clarity.

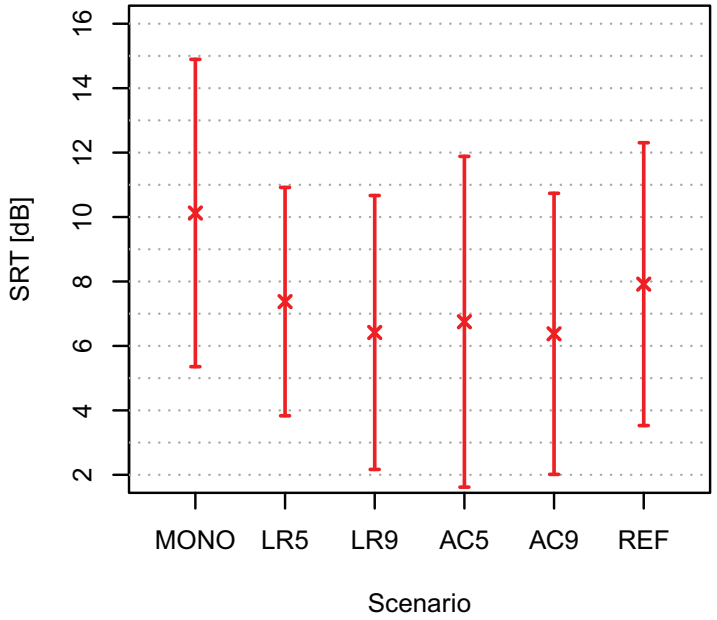


Figure 6.7: Test C results: Speech Recognition Threshold (SRT) in decibels in the six test scenarios. Marginal means of the SRT with 95% confidence intervals are presented.

6.5 Comparison of the results in tests A, B, and C

6.5.1 On the comparability of the results

Although the general test procedures were the same in all three tests, there were some minor differences. The effect of these differences must be understood in order to compared the results between tests. The differences in the initial SNR of the adaptive track very likely had no significant effect, because of the adaptive testing method. The test subject orientation (facing 0° in test A and C, facing -60° in test B) probably had an effect on the test subjects' hearing performance, but the effect was constant through the scenarios. Thus, when using the SRT-scores from a given test relative to the reference scenario of the same test, the results are comparable between the three tests.

6.5.2 The effect of direct-to-reverberant ratio in the listening position

Figure 6.8 shows a comparison of all the six scenarios where five or nine loudspeakers were used. The mean errors in SRT compared to the reference scenario of respective test are presented. The mean error of a scenario was calculated as the mean of the individual SRTs that were each normalized to the individual SRT measured in the reference scenario. Figure 6.8 shows that increasing the DRR in the listening position decreases the speech intelligibility and thus increases the error in the SRT.

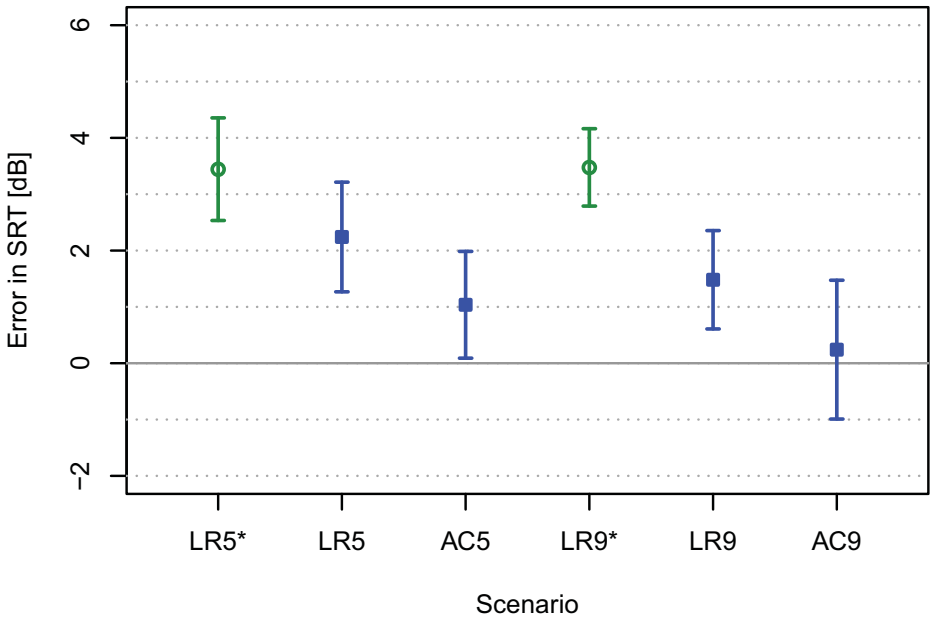


Figure 6.8: Comparison of the results in tests A and B: the effect of direct-to-reverberant ratio with five and nine loudspeakers. Marginal means of the errors in SRT compared to the respective reference scenario are presented with 95% confidence intervals. Data from test A in green circles and data from test B in blue squares.

In the test scenarios, DRR was altered by two factors: the amount of reverberation in the test room and the distance of the loudspeaker from the listening position. Scenarios LR5* and LR9* had the lowest DRR, as the loudspeakers were located in a distance of 2.3 m from the test subject. In scenarios LR5 and LR9, the room reverberation was the same as in LR5* and LR9*, but loudspeakers were located in a distance of 1 m, leading to higher DRR. In scenarios AC5 and AC9, there was no reverberation as the test were conducted in anechoic chamber, thus leading to the highest DRR.

To gain more insight of the relation between test room DRR and the error in SRT, the DRR was measured in the listening room with loudspeaker distances of 1.0 m and 2.3 m. Results are presented in Table 6.5. The measurements procedures are described in detail in appendix D.

Table 6.5: Direct-to-reverberant ratio (DRR) in the listening room prototype setup for loudspeaker distances of 1.0 meter and 2.3 meters.

Distance to loudspeaker	1.0 m	2.3 m
DRR	8 dB	1 dB

Table 6.5 shows that there is approximately 7 dB of more direct sound present in scenarios LR5 and LR9 than in LR5* and LR9*. This is logical, since for a point source, halving the sound source distance in free field increases the sound pressure by 6 dB [73]. In an ideal anechoic chamber, DRR is infinite. Although there were some reflecting objects in the anechoic chamber used, the reverberant energy can be considered very small and thus the DRR very high.

6.5.3 The effect of the number of loudspeakers

In addition to the effect of DRR, Figure 6.8 shows that the mean error in SRT was slightly smaller in the the nine-loudspeaker scenarios than in the five-loudspeaker scenarios. A major reason for this is assumed to be the uneven distribution of the loudspeakers in the five-loudspeaker setup. Especially when the test subject was in the rotated position (-60° azimuth), the five-loudspeaker setup becomes asymmetrical in left-right direction, whereas the nine-loudspeaker setup remains more symmetric in this sense.

Increasing the number of loudspeakers further from nine did not decrease the error in SRT significantly. Namely, Figure 6.3 and the analysis in Section 6.2.4 showed that using 13 loudspeakers did not result in significant difference in the SRT compared to the scenarios with five or nine loudspeakers. However, this was examined only with the loudspeaker distance of 2.3 m.

6.5.4 The effect of test subject hearing performance

Figure 6.9 shows a comparison of the results from tests B and C. Similarly to Figure 6.8, the mean errors in SRT compared to the reference scenario of respective test are presented. Again, the mean error of a scenario was calculated as the mean of the individual SRT-scores that were each normalized to the individual SRT measured in the reference scenario.

Although normal hearing test subjects had the best intelligibility in the reference scenario, it seems that for the hearing instrument users the intelligibility is slightly better in the DirAC-reproduction scenarios. However, in the test subjects group of test C the differences are quite small and the confidence intervals quite large. In both groups, the MONO-scenario produces the worst intelligibility and the largest error compared to the reference. All in all, there was no significant differences between any scenarios in test C.

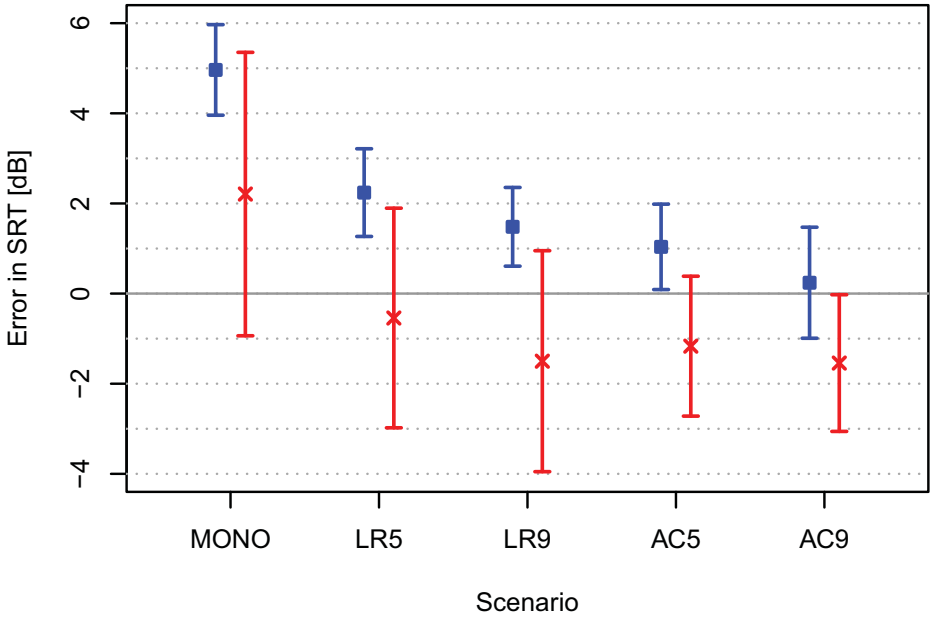


Figure 6.9: Comparison of the results in tests B and C: the effect of the test subject hearing performance. Marginal means of the errors in SRT compared to the respective reference scenario are presented with 95% confidence intervals. Data from test B in blue squares and data from test C in red crosses.

6.6 Reliability of the results

The reliability of the results can be considered generally good. There is still a few possible sources of error.

Most importantly, there was considerable amount of noise in the data, as could be seen from the individual SRT-scores from all tests (Figures 6.2, 6.4, and 6.6). That is, not all individual tracks clearly followed the overall trend. Visual inspection of the adaptive tracks in the tests revealed that not always clear adaptation happened: in some cases, the track wandered up and down aimlessly. There are several facts related to the word lists that could have caused this. First, the word lists used are considered a bit old-fashioned by the author. That is, some of the words were uncommon, of old style, or ones that can be considered colloquial. These findings were also agreed by many of the test subjects during informal discussion after the test, in which the test subjects were asked: ”how did you like the test speech material?”. It is obvious that the word lists should consist only of words which everybody knows and understands, and now there was a few exceptions

degrading the balance of the lists. Second, the speech corpus was quite small. The noise in the results could probably have been decreased by using a larger corpus. Namely, with a larger number of inter-compatible word lists each scenario could have been conducted with several interleaved lists. A longer word lists could also have improved the adaptation. In tests A and B an adequate number of test subjects was used to average the noise out of the data. In test C the number of test subjects was unfortunately lower, but probably fair enough for the analysis required.

Another issue is the comparability of the SNRs. The SNR between masker and test speech in the listening position was carefully measured using the same method for all scenarios. This ensured that the results are comparable between scenarios. Small variation to the effective SNR has probably been caused by the fact that test subjects were not denied to move their head during the test. Still, based on visual observations during the test procedures, the effect of this is quite small.

In contrast, there may be error in the absolute value of the SNR. This is due to the difficulty of defining the SNR between a masker and reverberant speech, or to be precise, of defining what part of the speech is actually the desired signal. That is, in reverberant speech, the direct sound and early reflections are desired and increasing their level increases the intelligibility. On the other hand, late reflections and excessive reverberation can instead decrease the intelligibility. However, in these tests, the absolute value of SNR is not of interest: the test method is based on relative differences in the SRT obtained in different scenarios. Thus, in these tests, all energy of the test speech was regarded as the desired signal.

In test A, the fixed order of scenarios could have caused learning effects, if listening to the first scenarios gained the test subjects performance in the last scenarios. However, this did not probably have significant effect to the results. First reason for this is that the test procedures represented nothing more than communication in noise, which is quite an everyday-task. Secondly, different word lists were used in different scenarios, and they were randomized between scenarios. Thus, benefiting from memorizing the words was not possible. Regardless, in tests B and C, the scenario order was balanced with a latin square to avoid order-related effects.

Chapter 7

Discussion

This chapter concludes and interprets the outcome of the listening tests and evaluates the proposed DirAC-based sound-field audiometry system. Suggestions are made for a clinical implementation of the concept, and possible future work is discussed.

7.1 Outcome of the listening tests

The outcomes of the listening tests reported in Chapter 6 can be concluded as follows. First, both the number of loudspeakers and the direct-to-reverberant ratio has an effect on the SRT. When nine loudspeakers were used, there was no significant difference between the SRTs obtained in the reference environment and the corresponding DirAC-reproduction in the anechoic prototype setup or the listening room prototype setup with one-meter loudspeaker distance. Decreasing the DRR decreased the SRT-scores and thus increased the error compared to the reference environment. Using an asymmetrical five-loudspeaker setup led to higher error compared to a nine-loudspeaker setup, but was still found to be valid in anechoic conditions. Second, compared to monophonic reproduction, the DirAC-reproduction resulted in smaller error in SRT. This motivates the use of several loudspeakers instead of one. Finally, the test subject group consisting of hearing instrument users did not perform in significantly different ways in the tested scenarios. This confirmed that the psychoacoustic assumptions of DirAC were valid also with listeners with non-normal hearing.

7.2 Evaluation of the DirAC-based SFA system

7.2.1 Validity

The outcomes of the listening tests can be interpreted as follows. First, the proposed concept of DirAC-based sound-field audiometry is valid at least in terms of speech intelligibility assessments. This is due to no significant indifferences in the SRTs measured in a reference "real life" scenario and its DirAC-reproduction equivalent. This validity is met when using a symmetrical setup of nine loudspeakers in a room with a DRR of 8 dB or higher in the listening position. The validity is ensured only with the particular audio

used in the listening tests. Thus, no explicit conclusion can be made on the validity when other sound scenes are reproduced.

Valid results might be achievable with somewhat looser requirements, for example with a lower number of loudspeaker – given that the loudspeakers are evenly placed – or with a lower DRR. However, a conventional 5.1-setup was found to be insufficient even with DRR of 8 dB as well as was DRR of 1 dB with nine loudspeakers.

7.2.2 Advantages and drawbacks

Compared to existing methods for sound-field audiometry, the proposed method has several clear advantages. First, no more than nine loudspeakers in the horizontal plane are needed for adequate reproduction concerning speech intelligibility assessments in realistic sound scenes. Second, due to the use of parametric audio coding, the number of loudspeakers and their positioning are arbitrary as long as the minimum requirements are met. Third, the system is capable of reproducing real sound scenes including the associated room acoustics, in which anechoic test speech material can be augmented. In contrast, many of the current methods are based on simulations or use synthesized audio material. Using real recorded environments enables more realistic testing. Finally, the system is modular as different background sound scenes, room acoustics, and anechoic speech corpuses can be combined freely.

The proposed method has also some drawbacks. The system must be calibrated carefully to obtain reliable results. That is, the SPL of the masker and test speech must be measured and adjusted carefully to ensure the correct SNR in the test. This is somewhat a common drawback in all sound-field audiometry applications, but is emphasized when real recorded audio is used.

7.2.3 Suggestions for clinical implementations

Suggestions for a clinical DirAC-based SFA setup specifications are as follows.

The preferred arrangement of loudspeakers is a somewhat even and symmetrical arrangement in the horizontal plane at the ear-level of the listener. Nine loudspeakers are recommended.

The test room should have a relatively low reverberation time. Generally, the lower the better. The listening room used in the test of this thesis is an example of a suitable test room (see Section 5.3.3 for specifications). In that room, the loudspeaker distance of 1 meter resulted in a DRR of 8 dB. Following a listening room recommendation (such as ITU-R BS.1116 [35]) or the quasi-free sound field conditions [33] would be advantageous in the long run. That is, anchoring the requirements to an existing standard would make it easy to build new setups and maintain the comparability between them. However, the most important factor is the DRR. The test room and loudspeaker distance from listening position should be designed so that a DRR of 8 dB is achieved in the listening position. Preferably, DRR measurements should be done to validate the room for this purpose. Loudspeaker distance of 1 m is suitable. Distances below this are problematic, while the closer the loudspeakers are, the bigger is the relative change in the loudness when head

is moved. On the other hand, with much higher loudspeaker distances it may be hard to achieve a high enough DRR.

The audio material used should be selected depending on what kind of situations and scenarios are wanted to be tested. In choosing the masker a compromise is made between the realism of the scene and reliability of the test. The speech corpus to be used should be large enough and have equal difficulty thorough the test. All audio material used should be high-pass filtered with a cut-off frequency equal and higher than the Schroeder frequency of the test room to prevent the effect of room modes.

When implementing the proposed SFA setup in a clinic, the SNR between masker and test speech must be defined carefully. This is essential if results are wanted to be comparable for example between different clinics. On the other hand, when conducting relative measures, such as functional gain of a hearing instrument or test similar to which was done in this thesis, the absolute SNR needs not the be strictly defined. That is, in these cases the error is subtracted in the comparison. According to [8], the energy in the first 80 ms could be regarded as the desirable sound and the energy after that as the undesired sound. This could be one approach to use for dividing a highly-reverberated test speech to actual signal energy and masker energy. However, the extent in which the reverberant sound acts as a masker should be carefully analyzed. That is, if a two-syllable test word is used, a three-second reverberation decay is mostly present after the word has ended and is obviously not masking anymore. When using sentence material, the situation can be somewhat different as the reverberation of previous words mask the following ones.

7.3 Suggestions for further work

The validity of the concept with some different masker and test speech material is unsure. Thus, one aim for further research could be to repeat the listening tests of this thesis with some other reference scene. The reference sound scene used in the tests of this thesis had relatively low reverberation. Using a scene with higher reverberance might have some effect on the validity of the system. A guess by the author is that using such highly reverberant reference scene would probably loosen the DRR-criteria of the test room. Namely, in this case the additional room effect applied by the test room would be smaller compared to the original reverberation and thus have smaller effect on the speech intelligibility. Also, the use of sentence test speech material together with a percent-intelligibility test procedure could be considered. In that case, the masker stationarity criterion might not be as strict as when using single words and a SRT procedure.

Only speech intelligibility assessments were discussed in this thesis. However, the concept of DirAC-based sound-field audiometry might be extendable to other audiometric measurements as well, but these applications should be validated with listening tests depending on motivation. Possible applications could be such as hearing-based orientation tasks in reproduced sound scenes.

Adding a visual display to the system would add more realism. This would allow the visual cues that are naturally present in real environments (e.g., a change to lip-read), which would bring the results even closer to the real-life performance. This could be especially useful when assessing the overall communication performance of hearing-impaired individuals, while they may gain significant benefit from visual cues. Adding a visual

display could further broaden the application area and enable for example attention tests in a virtual classroom.

Future work could also include implementing the DirAC-based SFA system as a standalone software for the use of healthcare. In this case, capturing a set of suitable sound scenes would be also required. For the system to be accepted for diagnostic use, a prototype setup should be built in a clinic and further validation done with different sound scenes and larger number of test subjects. This could probably be implemented most efficiently in a multidisciplinary project involving both physicians and engineers.

Chapter 8

Conclusion

This thesis investigated the use of a parametric spatial audio reproduction technique called Directional Audio Coding in the field of hearing diagnostics. The motivation for this research was given in Chapter 1. The theory part, consisting of Chapters 2–4, explored the background of the topic: these chapters discussed the principles of sound and hearing, spatial audio technology, and technical audiology with an emphasis on sound-field audiometry.

In the experimental part, consisting of Chapters 5–7, the concept of a sound-field audiometry system based on Directional Audio Coding was motivated, designed, and validated. A prototype setup of the system was built and a MATLAB-software was written for testing the concept. Additionally, a reference sound installation representing a real environment was built for comparison. The concept was validated with listening tests using normally-hearing individuals and users of hearing aid and/or cochlear implant as the test subjects. The listening tests revealed that with an adequate number of loudspeakers and adequately controlled acoustics in the test room, the same speech intelligibility could be achieved in the reproduced scene as in the reference scene. Based on these findings, suggestions were made for a clinical implementation of the system.

The proposed method offers a platform for various audiometric tests in real sound scenes with external speech material augmented to the associated room acoustics. The proposed method is more compact and flexible than most of the existing methods in terms of system modularity and the loudspeaker setup needed. Consequently, the proposed method could be easily applied to clinical purposes.

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Appendix A

SPS200 Compensation filter

A compensation filter was implemented to flatten the magnitude response of the Soundfield SPS200 A-format microphone response. The microphone consists of four capsules, each of which have their own magnitude responses. These four responses were measured in [61] for the same microphone as was used in the work of this thesis. These responses were analyzed by the author and it found out that the four responses were very similar to each other. Thus, a common compensating filter for all the four A-format channels were formulated from the on-axis impulse response of the left–front-capsule. The compensation filter was calculated as the inverse of the original impulse response. The microphone response has a roll-off below 90 Hz and above 18 kHz. These frequency areas were low-pass filtered after the inversion to avoid overshoot in the inverted response. Figure A1 presents the original magnitude spectrum of the microphone (the black dashed line) and the magnitude response of the compensation filter (the red solid line).

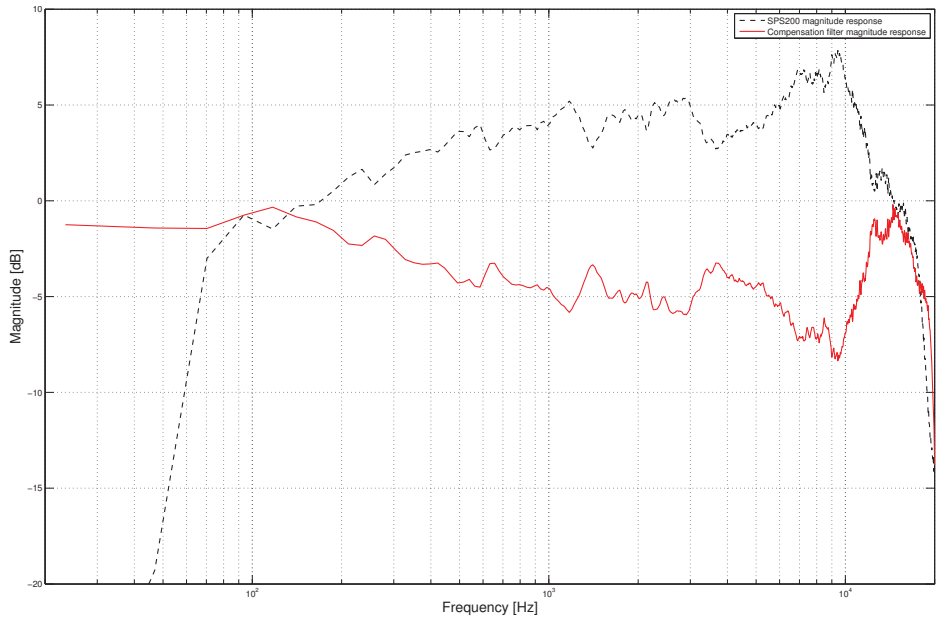

Figure A1: Magnitude responses of the Soundfield SPS200 microphone and the calculated compensation filter.

Appendix B

Reverberation time measurement details

Reverberation time (RT) was measured in the reference environment and in the listening room following the standard ISO 3382 [30]. Interrupted pink noise was used as the excitation signal. Total of 12 measurements were done in both rooms, namely in two loudspeaker locations, each with three microphone locations, and each with two repetitions.

The noise was generated in MATLAB and reproduced with an active loudspeaker (Genelec 8030A). Measurements were done with a microphone (B&K, type 4192 capsule and type 2669 preamplifier) and a conditioning amplifier (Nexus) connected to a computer (Apple Macbook) via an audio interface (MOTU Traveler mk3).

Analysis was done in MATLAB, with a script utilizing Schroeder backward integration [76]. T30 procedure was used in the analysis. That is, the time for the sound to attenuate from -5 dB to -35 dB was analyzed and the respective value for the 60 dB attenuation was calculated by doubling the RT30 value. The analysis was done in nine octave bands: 63 Hz to 8 kHz. The final value of RT in each octave band was calculated as the average of the 12 measurements. Finally, an average value for RT was calculated as the average of the values of octave bands 500 Hz and 1 kHz.

Measurements from the 31 Hz octave band were not reliable due to the limited frequency response of the loudspeaker used. This octave band was however not of interest in this thesis, while all audio used in the tests was high-pass filtered with 100 Hz cut-off frequency.

Appendix C

Post-hoc analysis tables

C.1 Test A

Table C1 shows the output of Dunnett's modified Tukey-Kramer pairwise multiple comparison test [20] for test A results. In each row, a pair comparison is made to discover whether the difference in the mean SRT in the two compared scenarios is significant or not. The last column indicates the significance.

Table C1: Post-hoc analysis results for test A indicating the significance of the differences between scenarios.

	Diff	Lower CI	Upper CI	Significant difference?
LR5*-MON*	-1.48	-3.43	0.48	no
LR9*-MON*	-1.44	-3.45	0.57	no
LR13*-MON*	-2.11	-3.86	-0.36	yes
LRA*-MON*	-4.49	-6.42	-2.56	yes
REF-MON*	-4.92	-6.77	-3.07	yes
LR9*-LR5*	0.03	-1.83	1.90	no
LR13*-LR5*	-0.63	-2.21	0.95	no
LRA*-LR5*	-3.02	-4.80	-1.24	yes
REF-LR5*	-3.44	-5.13	-1.76	yes
LR13*-LR9*	-0.67	-2.32	0.98	no
LRA*-LR9*	-3.05	-4.89	-1.21	yes
REF-LR9*	-3.48	-5.23	-1.72	yes
LRA*-LR13*	-2.38	-3.93	-0.83	yes
REF-LR13*	-2.81	-4.25	-1.36	yes
REF-LRA*	-0.43	-2.09	1.23	no

C.2 Test B

Table C2 shows the output of Dunnett's modified Tukey-Kramer pairwise multiple comparison test [20] for test B results. In each row, a pair comparison is made to discover whether the difference in the mean SRT in the two compared scenarios is significant or not. The last column indicates the significance.

Table C2: Post-hoc analysis results for test B indicating the significance of the differences between scenarios.

	Diff	Lower CI	Upper CI	Significant difference?
LR5-MONO	-2.72	-4.66	-0.79	yes
LR9-MONO	-3.48	-5.77	-1.19	yes
AC5-MONO	-3.93	-6.04	-1.81	yes
AC9-MONO	-4.72	-7.12	-2.33	yes
REF-MONO	-4.96	-7.27	-2.65	yes
LR9-LR5	-0.76	-2.62	1.10	no
AC5-LR5	-1.20	-2.84	0.43	no
AC9-LR5	-2.00	-3.98	-0.02	yes
REF-LR5	-2.24	-4.12	-0.36	yes
AC5-LR9	-0.44	-2.49	1.60	no
AC9-LR9	-1.24	-3.57	1.09	no
REF-LR9	-1.48	-3.73	0.76	no
AC9-AC5	-0.80	-2.96	1.36	no
REF-AC5	-1.04	-3.10	1.03	no
REF-AC9	-0.24	-2.59	2.11	no

Appendix D

Direct-to-reverberant ratio measurement details

Direct-to-reverberant ratio was measured in the listening room prototype setup with two loudspeaker distances, namely 1 m and 2.3 m. The measurements and analysis were done in MATLAB. A logarithmic sine sweep was reproduced in the loudspeaker position and recorded in the listening position. The sweep was reproduced with an active loudspeaker (Genelec 8030A) and recording was done with a microphone (B&K, type 4192 capsule and type 2669 preamplifier) and a conditioning amplifier (Nexus) connected to a computer (Apple Macbook) via an audio interface (MOTU Traveler mk3).

From the sweep, an impulse response was calculated. The impulse response was windowed with a rectangular window so that direct and reverberant part were separated. The time delay between the direct sound and the first reflection was calculated to ensure correct location for the separation. Finally, the sum of squares was calculated for both parts and the DRR was calculated as logarithm of their ratio. This analysis was done separately for the two loudspeaker distances.